

(NASA-CR-150865) A STUDY OF DATA CODING
TECHNOLOGY DEVELOPMENTS IN THE 1980-1985
TIME FRAME, VOLUME 2 Report, Sep. 1977 -
Dec. 1978 (Mississippi State Univ.,
Mississippi State.) 99 p HC A05/MF A01

N79-12774

Unclas

G3/61 39017



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A STUDY OF DATA CODING TECHNOLOGY DEVELOPMENTS
IN THE
1980-1985 TIME FRAME

Volume II of II

by
Frank M. Ingels
Mohammad Mehdi Shahsavari



MSSU-EIRS-EE-79-1-VOL-II

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IN THE
1980-1985 TIME FRAME

by

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Submitted to

George C. Marshall Space Flight Center
National Aeronautics and Space Administration
Marshall Space Flight Center, Alabama

Under

NASA Contract No. NAS8-31373
For the Period
September 1977 through December 1978

Volume II of II

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SUMMARY

The work statement items of this contract have been addressed in this Final report in the following manner:

- A) The objectives of this report and the source parameters of digitized analog data have been discussed in Chapters I and II.
- B) Different data compression schemes, and analysis of their implementation have been presented in Chapter III.
- C) Chapter IV presents bandwidth compression techniques for video signals.
- D) Results and conclusions have been presented in Chapter V including a best estimate of the coding schemes to be used in 1980-1985.

Appendix-I presents a chronological bibliography.

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CHAPTER I

INTRODUCTION

In recent years the demand for efficient and reliable digital transmission systems has been accelerated by increasing use of automatic data processors. The need has also increased for long range communications for global crop production forecasting, local weather and severe storm predicting, water availability forecasting, land use and monitoring. These applications involve high data volumes, high transmission rates, the need for rapid throughput times and the necessity of handling images from satellites.

Two of the more serious problems in any high speed data transmission systems are: 1) occurrence of errors, and 2) the need for bandwidth compression.

The former problem, that of occurrence of errors will result in the need for error correcting coding, unless the data channel is highly reliable. Some consideration has been given to efficient error correcting coding techniques, and is presented in the open literature.

The emphasis of this report is on the later problem, that of efficient bandwidth compression techniques for voice, telemetry and video which will be hardware feasible in the 1980-1985 time-frame.

Most data sources (voice, telemetry, video) generally are redundant in nature. Redundancy has been defined by Shannon as, "That

fraction of a message or datum which is unnecessary and hence repetitive in the sense that if it were missing the message would still be essentially complete, or at least could be completed." Removal of the redundancy from the source will reduce the bandwidth requirement, and is called "data compression" or "redundancy reduction".

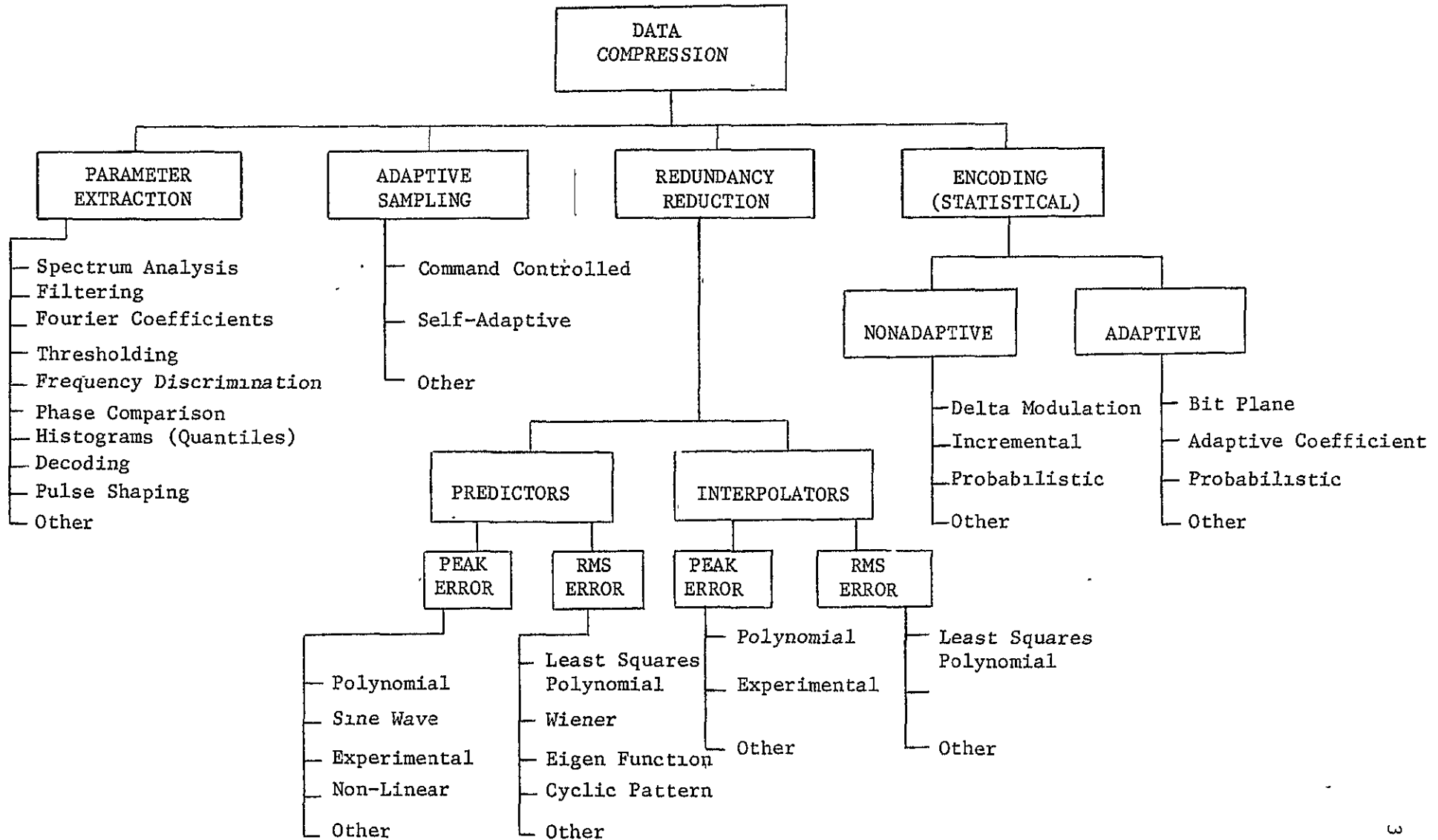
Knowing the best data redundancy reduction scheme to use for a particular application depends on knowing the data source characteristics. A characterization of source parameters of analog data after digitization is discussed in Chapter II.

Basic data compression techniques have been applied for many years in the processing of data by human analysts through the seeking of significant changes in data. Various data compression techniques (see Table I) are discussed and categorized in Chapter III.

Image coding for monochrome and color video links are discussed in Chapter IV. Techniques for reduction of video data by coding will depend upon the source data statistics. As in the case with any efficient data reduction encoding scheme, efficient run-length encoding requires a prior knowledge of the data source probability distribution. A detailed study of the average code word lengths versus the encoded word lengths for run-length encoding of sources using Huffman coding with Geometric, Poisson, Binomial and Modified Bionomial source distributions has been made and is reported in [33] and [36].

Results and recommendations are given in Chapter V. A chronological bibliography of the references used is presented in Appendix I.

TABLE I
CLASSIFICATION OF DATA COMPRESSION MODELS



CHAPTER II

SOURCE PARAMETERS OF DIGITIZED ANALOG DATA

Introduction

Digitizing an analog source (voice, video, telemetry data) involves sampling the time - continuous waveform and then quantizing these samples, as shown in Figure 1. Analog sources can be digitized in a way so as to minimize the source redundancy and bit rate.

Let m denote the number (alphabet size) of different symbols available from either a digitized analog source or a natural digital source, occurring at a rate $1/T_s$. Since m symbols can be represented by $k = \log_2 m$ binary symbols, the source bit rate R_s is

$$R_s = \frac{1}{T_s} \log_2 m = \frac{k}{T_s} \frac{\text{source bits}}{\text{second}} \quad (1)$$

where

R_s source rate in bit per second

T_s Time per source symbol

k number of binary digits per word or symbol.

When binary representation is actually used, as in digital computers, one can either consider $m = 2$ with T_s equal to the bit interval or consider $m = 2^k$ with T_s equal to the k - bit word length.

A digital information source is said to possess redundancy if either:

- I) The symbols are not equally likely
- II) The symbols are not statistically independent.

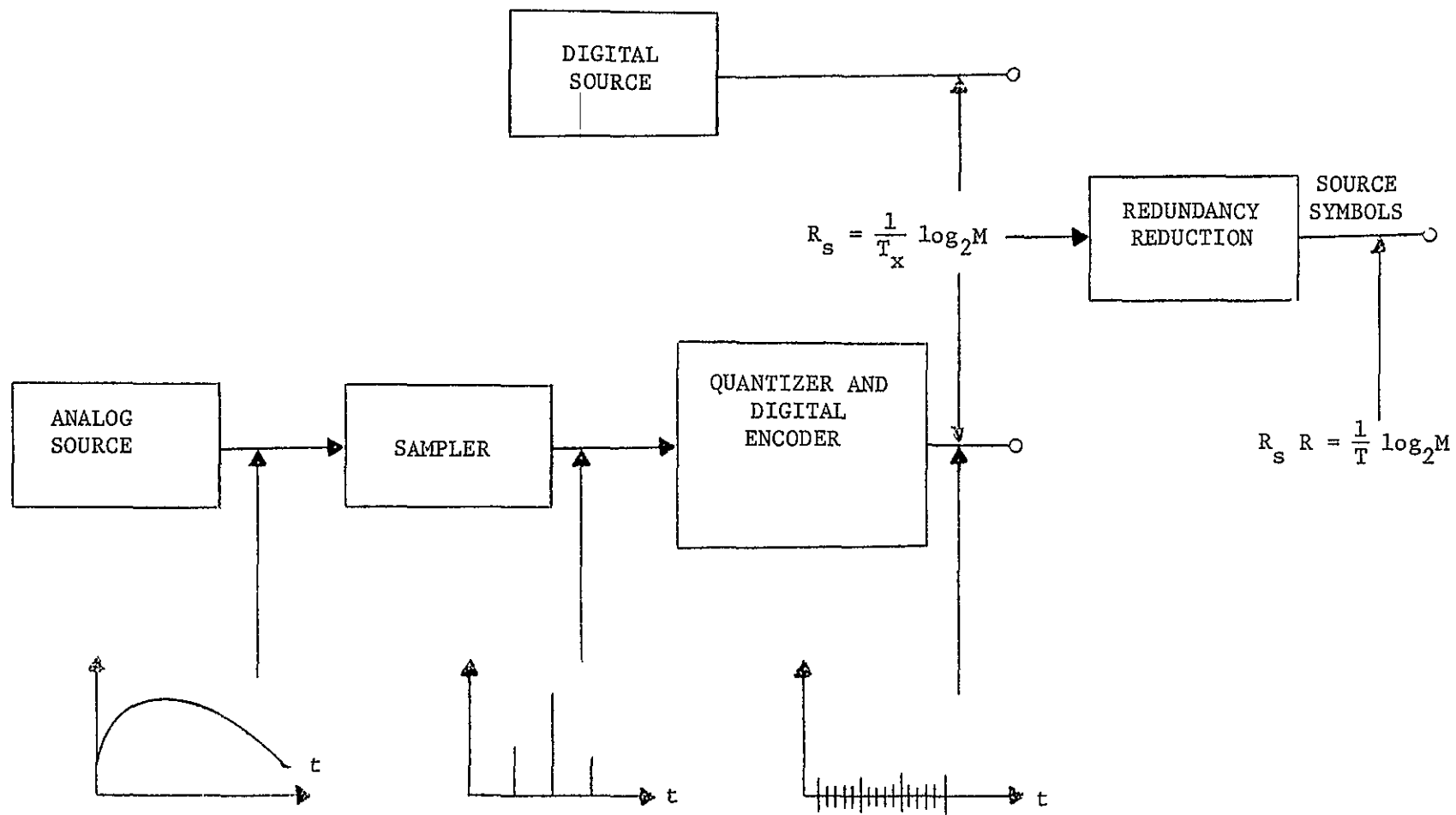


Figure 1. Elements in Source Encoding.

Source encoding can reduce the rate of source symbols if either of these conditions exists. If the source symbols are statistically independent, the source is said to have ZERO MEMORY since each symbol is selected without influence from any previous symbols.

Conceptually, one can reduce the source bit rate down to the theoretical information rate or entropy rate. For a source with zero memory, it is known classically that the average amount of information bits per second from an alphabet x_1, x_2, \dots, x_m is given [44] by

$$\begin{aligned} R_I &= - \frac{1}{T_s} \sum_{i=1}^m P(x_i) \log_2 P(x_i) \\ &= \frac{\text{Symbols}}{\text{Seconds}} \times \frac{\text{Information bits}}{\text{Symbol}} \end{aligned} \quad (2)$$

where

R_I information rate in information bits per second

T_s seconds per symbol

m alphabet size

$P(x_i)$ occurrence probability of symbol x_i

The summation term is the classical entropy, which evaluates the information bits per symbol. Note that the source bit of (1) is a binary representation or a unit of measure; the information bit of (2) is theoretical information content.

For a source with memory, the equation corresponding to (2) involves conditional probabilities. Usually the encoding procedures which exploit both nonequally likely and memory redundancy are AD-HOC procedures. As an example, the picture phone signal uses frame-to-frame differential encoding for memory reduction and Shannon-Fano encoding of the resultant differential signals [38].

When a redundancy reduction technique is used, then final source rate R should be smaller than R_s of (1) and greater than the theoretical R_I of (2). Defining the final source rate in terms of m symbols per interval T :

$$R_s \geq R = \frac{1}{T} \log_x m \geq R_I \text{ b/s}$$

where

R = source bit rate in bits per second. The source rate R is equal to the information bit rate only if the m symbols are equally likely and statistically independent ($R \geq R_I$).

Redundancy reduction for analog sources usually involves reducing the memory redundancy among the sequential samples. The objective is to minimize the eventual bit rate while providing the necessary quality.

Before reviewing digitization techniques, one should note the role of quantization noise. Any quantizing of a continuous quantity produces an inevitable uncertainty (quantization noise) about the precise value within the QUANTIZATION INTERVAL or step size. The signal's total amplitude span is divided into m levels or quanta. Increasing m reduces the quantization noise, but increases the symbol bit rate needed to represent the sample. Note that specifying m is equivalent to specifying or defining the tolerable distortion in the digitized source. Hence m is chosen in accordance with answering the question: What precision in amplitude level is required or desired by the user? The size of m directly affects the source bit rate, but not the redundancy due to memory. Generally a 7-bit representation ($m = 2^7 = 128$) for PCM yields excellent quality; 6-bit ($m = 64$) is sometimes used when the quality required is less stringent.

When varying signal volume (changing amplitude span) is an issue, a fixed or a time varying compression curve (nonlinear input-output amplitude relation) is used to make efficient use of the m levels.

While quantization noise usually represents signal degradation imperceptible to the user, additional degradation occurs if (channel) errors are made during the digital transmission. Typically one chooses the m so that the quantization noise is barely imperceptible to the user and then uses a combination of signal power, encoding, and repeater spacing to keep the channel errors negligible.

In the following, a brief review of existing analog digitalization encoding methods is given. Note the accompanying bit rate.

PCM

The classical analog digitalization is pulse-code modulation (PCM)*. In PCM analog signal is first sampled at a rate exceeding twice (usually 5 times) the highest significant frequency present in the signal (Nyquist rate), and each sample is quantized to one of a finite number of (m) levels. Each such finite level is a digital symbol and is represented by a digital alphabet. Traditionally, PCM is associated with binary signaling because most such systems used binary. The digital signals are then sent sequentially.

For a nominal 4KHz voice signal, an m of 128 ($k = 7$) results in a source bit rate of 56Kb/s ($[7 \text{ information bits}] \times 8 \text{ k samples/}$

*The terminology PCM is justified as follows. The baseband digital signal is a pulse of some type. Coding here refers to expressing the m level quantized samples via a sequence of binary symbols. The term modulation in PCM is somewhat a misnomer; the coded pulse can be sent over a channel using a wide variety of modulation. No particular channel modulation is implied by PCM.

second), for $k = 6$, $m = 64$, 48Kb/s. Considering a 5MHz bandwidth for commercial television (video) signals, a 70 mb/s rate results in $m = 128$ with PCM. As seen PCM requires a bandwidth of more than 10 times that which analog transmission would require. Theoretically, PCM is relatively inefficient for digitizing most analog sources; its symbol rate far exceeds the actual information (entropy) rate of the source. $R \approx R_I$ if and only if the analog source had: 1) uniform amplitude statistics, and 2) a flat (or white) power spectral density versus frequency. Digitization methods that improve the efficiency over PCM also increase the cost of the terminal.

DPCM

In differential PCM (DPCM) the error between the current sample and a predicted value is PCM encoded and transmitted. The prediction computation is time invariant, and a basic tradeoff exists between the increasing use of sampling rates above the Nyquist rate and the m value of quantization levels used for the error signal. For speech (4KHzBW) DPCM is applied in an interframe fashion, and from sample to sample for video. Often a variable length code (Shannon-Fano or Huffman) is applied to DPCM values. In picturephone video, inter-frame DPCM is used along with a range of other encoding [34] techniques (including Shannon-Fano) to effect a 1.544-mb/s rate for the 1MHz analog signal.

Figure 2 shows that, for a fixed bit rate, DPCM would give a 14-db improvement over average standard PCM for video scenes and 16.8 db improvement over standard PCM for video. The advantage of DPCM can also be expressed in terms of bit rate. For a given S/N ratio,

DPCM gives a reduction in bit rate over standard PCM of about 18 megabits (2 bits/sample). Since the sampling rate for DPCM and PCM is assumed to be twice the bandwidth or 9 MHz, these curves in Figure 2 are actually defined only at multiples of 9 megabits [9].

DM

Delta modulation (DM) can be considered as the simplest form of DPCM. Now the predictor simply integrates (with some variable gain) the past history and attempts to stay close to the actual waveform. Only a binary error signal is transmitted in DM, depending on whether the integrated prediction is higher or lower than the actual waveform at the sampling time, which now may be much larger than the Nyquist rate since there is only one binary symbol per sample. An adaptive DM (ADM) is formed if the binary step size (commanded by the binary transmission) is varied from time to time, according to local waveform needs. ADM can provide speech as low as 10 Kb/s; a rate of 20Kb/s provides good quality, while a 50 Kb/s rate affords excellent quality. Although ADM could be used for video encoding, [21] it is used almost solely for speech.

APD and LPE

These encodings are aimed at speech and involve more elaborate prediction algorithms than DPCM or DM. While in linear predictive encoding (LPE) the spectral envelope is predicted and the pitch is separately processed, the predictor in adaptive predictive encoding (APD) estimates both the pitch and the present spectral envelop. The error signal is transmitted in APD, and in LEP the

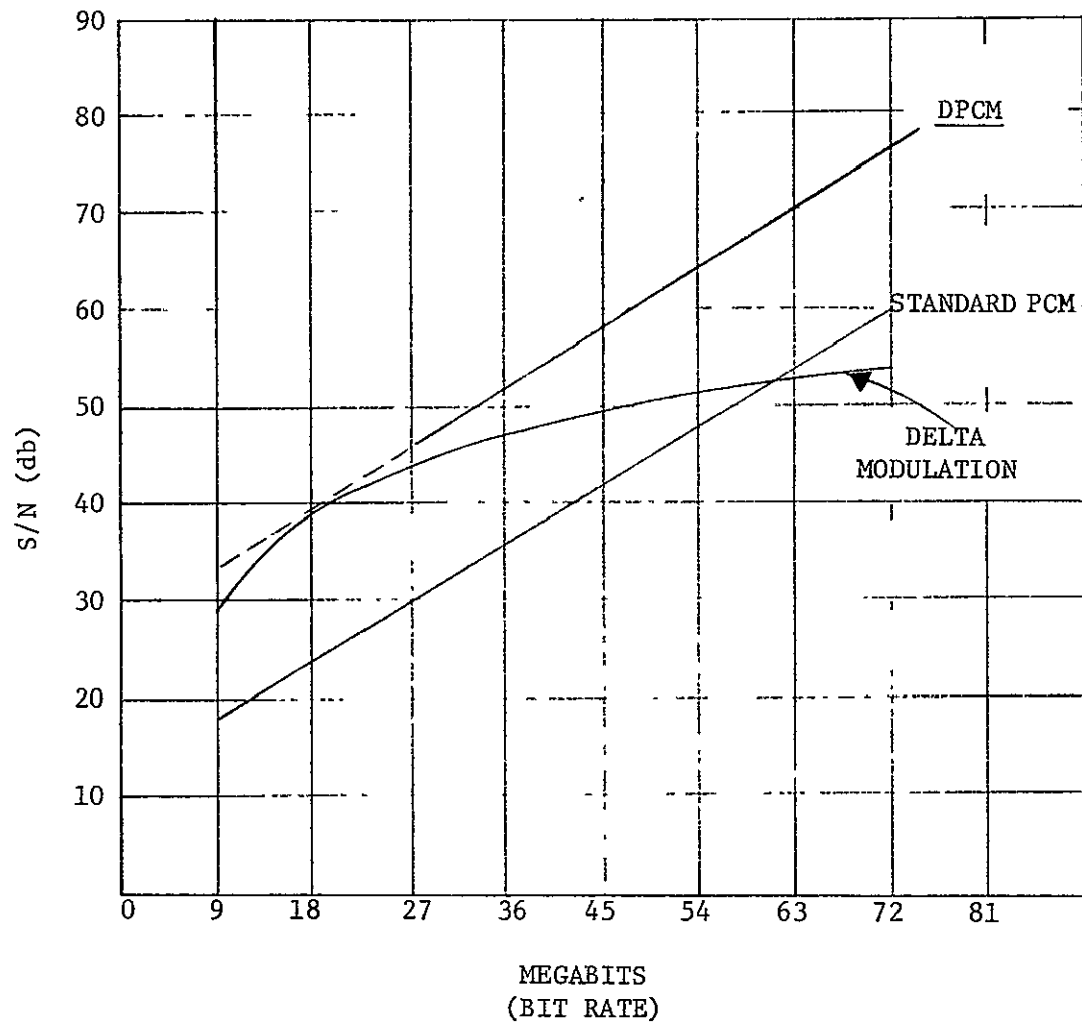


Figure 2. Comparison of S/N Versus Bit Rate For 4.5 MHz Television.

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error signal is characterized by sufficient parameters (pitch, voice characteristics) so that it can be recreated at the receiver. In laboratory experiments by B.S. Atal [30], speech of high quality was obtained at 7.2 Kb/s. At 4.8 and 2.4 Kb/s rates the quality is somewhat degraded, but better than any current competitors.

Time and Space Processing, Inc. [63], has marketed a digital telephone using linear predicting coding which for 4KHz bandwidth voice can compress the data to 2.4 Kb/s. The system is built to perform the operation at both ends of the line. This system can produce telephone grade voice at this rate. The price is \$14,950 and the whole processor measures 18.6 x 16.7 x 13.8 inches.

VOCODERS

Vocoders can be used to digitize voice signals. Although source rates as low as 1.2 Kb/s and .6K/bs are possible [32], the quality is poor and the complexity remains substantial. Chang, K. Un [80], has reported a simulation of synthetic speech encoding using a FORMANT VOCODER. The major advantage of the Formant Vocoder [32] is that, unlike other Vocoders, intelligible speech quality can be obtained at a transmission rate as low as 800 bits/s. This is possible because fewer transmission parameters are required than for other vocoders. The system algorithm presented draws heavily on the results of recent research in linear predictive coding (LPC). At transmission rates of 1.2 Kb/s, the synthetic speech quality is reasonably good; most of the utterances are intelligible and speaker recognizable. Increase of transmission rate to 1400 b/s resulted in slightly improved synthetic speech, but further increase of the bit-rate up to 1800 b/s

made little difference, indicating that quantization accuracy with the rate of 1400 b/s is a point of limiting returns.

In digital speech encoding systems, we have only a small amount of a prior knowledge of the statistics which, in addition, usually change with time [53]:

- I. The Long-Period mean level differs from speaker to speaker.
- II. At a given mean level, the instantaneous level changes because of variations in speech sounds.
- III. The correlations between successive samples change because of variations in speech sounds.

To overcome these problems of unknown statistics, adaptive quantization and adaptive prediction schemes must be used to convert the signal from analog to digital. Figure 3 [53], shows the optimum results reached with a three-bit quantization of the 2.3-s speech sample.

Left Curves: Optimum results using a fixed quantizer.

Right Curves: Optimum results using an adaptive quantizer.

Lower Curves: Prediction with a first-order predictor (one coefficient).

Upper Curves: Prediction with a high order predictor. Figure 4 shows the waveforms of the reconstructed signal of the quantization error for a 64 ms segment of speech. AQF = Adaptive Quantization Forward Scheme. In summary, Table II shows a listing of source parameters for voice and video sources.

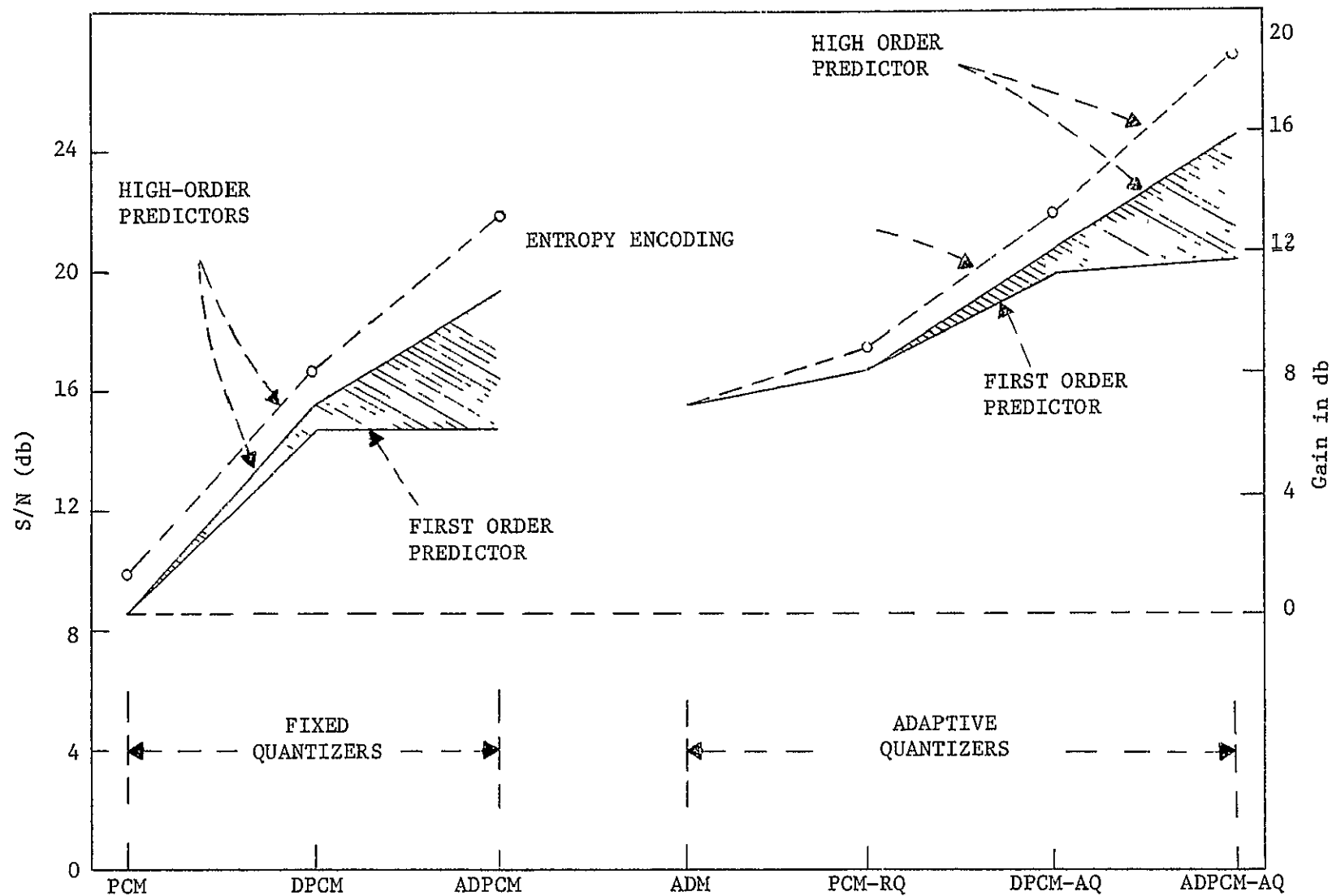


Figure 3. S/N and Gains for Different Three-Bit Speech Encoders.

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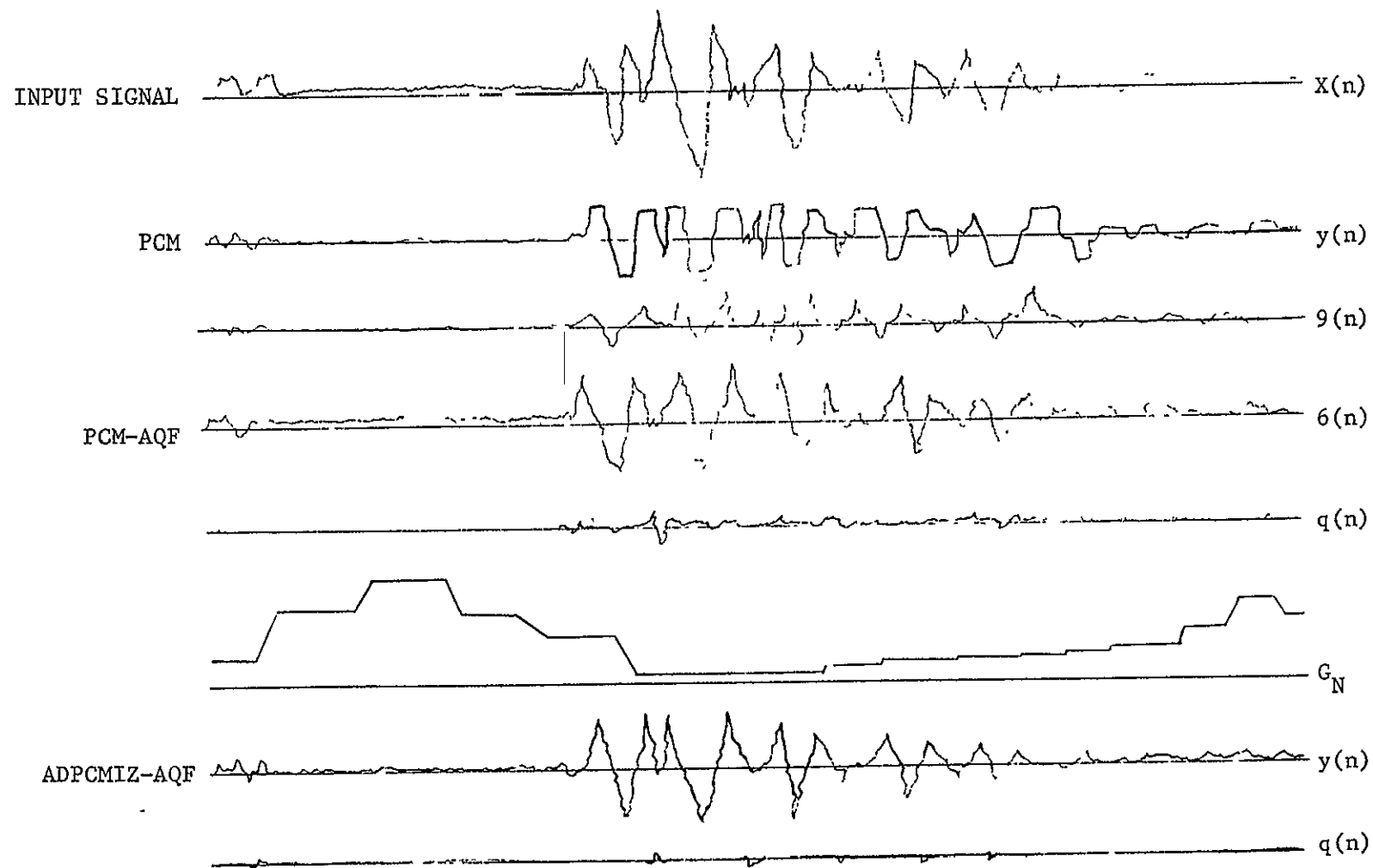


Figure 4. Comparison of Waveforms Over a Sequence of Input Samples, $y(n)$ = Sequence of Decoded Samples, $q(n)$ = Sequence of Quantization Errors, $G(n)$ = Sequence of Amplifier Gains.

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TABLE II
SOURCE PARAMETERS AFTER DIGITIZING

	<u>VOICE BIT RATES</u>	<u>VIDEO BIT RATES</u>
<u>PCM:</u>	(4 KHz BW)	(5 MHz BW)
m = 128, k = 7	56Kb/s	70Mb/s
m = 64, k = 6	48Kb/s	60Mb/s
<u>DPCM:</u>	40Kb/s (high quality)	(1 MHz BW)
	20Kb/s (medium quality)	1.544Mb/s (with Shannon- Fano Coding Technique)
<u>ADM:</u>		
(used solely for speech)	20Kb/s (good)	
	40Kb/s (excellent)	
	10Kb/s (poor)	
<u>APD and LPE:</u>		
voice and telemetry	7.2 Kb/s (high quality)	
	4.8 - 2.4Kb/s (degraded)	
<u>Vocoders:</u>		
	1.2Kb/s (high quality)	
	.6Kb/s (degraded)	

CHAPTER III

VARIOUS DATA COMPRESSION TECHNIQUES

Parameter Extraction

Parameter extraction is a technique that reduces the bandwidth required to transmit a given data sample by means of an information-describing irreversible transformation. These transformations are considered irreversible because, while they provide useful descriptions of the input signal, they so distort the signal that it is impossible to reconstruct the original waveform.

Parameter extraction represents the oldest and most widely used form of data compression. Signal conditioning devices providing reduction in information bandwidth are included in this category as are the extraction of the power spectrum of the signal and transmission of the amplitude of the spectral components (spectrum analyzers), phase comparators, and event recognition. Rather than send all the relevant signal data to be monitored for a significant event or critical situation, an event detector would detect the event and send extracted information such as time of the event or amplitude (peak detectors). These are a few examples of the virtually unlimited number of techniques that have been used or could be devised. It is generally considered good design to use a parameter extraction process wherever practical since, in many applications, it will reduce vehicle power, size, and weight requirements. In most circumstances, parameter extraction is complementary to and enhances other forms of data compression. Since the information desired from each

sensor may be different, such devices must be designed for the special parameter to be extracted and for the particular sensor characteristics.

Syndrome-Source-Coding

The conventional method [31] for using block error correcting codes to perform source encoding (or "data compression") is shown in Figure 5. The source output is treated as a received code word r , and a channel decoder for the selected code serves as the "source encoder". The channel decoder finds a codeword X close to r (in an appropriate sense) [4] and delivers as its output the information sequence u corresponding to this codeword. The sequence u is then the "compressed data." Thus, there are R compressed digits per source letter where R is the code rate, i.e. the ratio of the number of information digits to the total number of digits in codeword (for (n,k) binary block code $R = \frac{k}{n}$). At the user end, a channel encoder is used to convert u to the codeword X , and thus serves as the "source decoder".

Source encoding of the above type using linear error correcting codes has been shown to be efficient (by, T. J. Goblick) [4] for encoding memoryless symmetric sources under the Hamming distortion measure in which the average distortion is the average fraction of source digits erroneously reconstructed. For many "real" sources such as the output of measuring devices in space experiments, the information source is highly asymmetric with strong memory constraints. In such cases, the scheme of Figure 5, generally becomes inefficient since the codewords of simply implemented linear error correcting

codes do not well-approximate the likely set of source output sequences. For the conventional scheme of Figure 5, the more complex device, the channel decoder, is located at the source side, the simpler device, the channel encoder, is located at the user side.

An alternative method for using linear error-correcting codes to achieve data compression is diagrammed in Figure 6, and is called Syndrome-Source-Coding. The fundamental principle of syndrome-source-coding is that the source output is treated as a channel error pattern e , rather than as the received channel codeword. A syndrome-former*, the same complexity as an encoder for the code, serves as the "source encoder". The syndrome S , i.e., the pattern of parity-check failures, is then taken as the compressed data. There are $1/R$ compressed digits per source letter since the number of syndrome digits equals the number of redundant digits in a codeword. At the user end, an "error pattern-estimator" for the code, i.e. a device which produces a likely (in an appropriate sense) error pattern \hat{e} consistent with S , is used as the "source decoder". This device is the heart of the syndrome decoder for the code and dominates the complexity of the decoder in most cases.

Since the error patterns which are correctable by decoding schemes traditionally considered in channel coding studies tend to be asymmetric (e.g., for binary codes, the set of correctable error

*With any binary source, we associate the additive channel in which the source output form the error pattern, i.e. in which the received word is $r = X + e$, where X is the transmitted word, where e is the source output which is assumed to be statistically independent of X , and where the addition is component-by-component in $GF(2)$. A given syndrome decoder for a given linear code to be used on this channel would comprise two devices, the syndrome former that computes $S = rH^T$, where H is the parity-check-matrix of the code so that $S = rH^T = (X + e)H^T = eH^T$, and the error pattern estimator whose input is S and whose output is \hat{e} or e . The corresponding estimate \hat{X} of X is, of course, $\hat{X} = r - \hat{e}$.

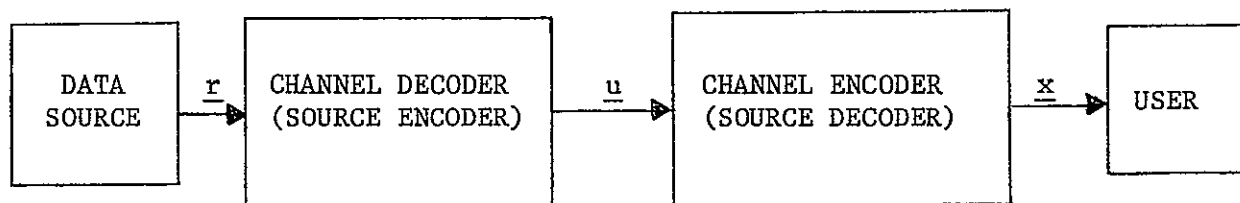


Figure 5. Conventional Method of Using Error-Correcting Codes For Data Compression.

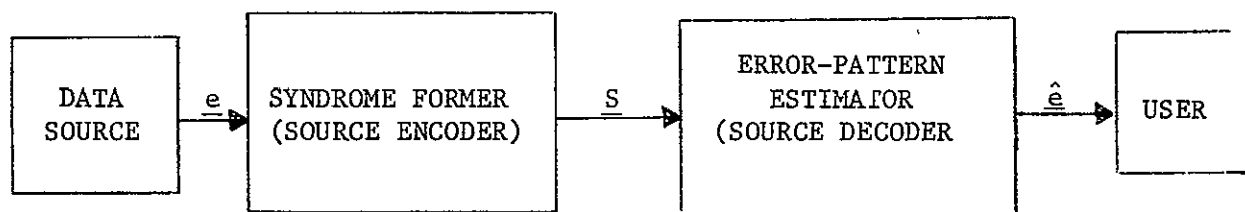


Figure 6. Syndrome Source Coding Method of Using Error-Correcting Codes For Data Compression.

patterns usually includes all sequences with a sufficiently small number of ones) and tend to match memory effects (e.g. the correctable errors often include all "bursts" in which all over in the error pattern are confined to some small span of consecutive positions), the syndrome-source-coding method appears well-suited to the compression of many real sources. In other words, the set of error patterns correctable by simply implemented decoders for known block codes seem to approximate well the "likely" set of source output sequences for many real sources. Moreover, Syndrome-Source-Coding has the advantage for many applications that the simpler device, the syndrome-former, is located at the source side while the more complex device, the error-pattern-estimator, is located at the user side.

A universal generalization of syndrome-source-coding (UGSSC) given by Ancheta [57] and performance of UGSSC is compared with Run-Length encoding with probability P of source emitting of a "one" for different lengths ($n = 15$, $n = 31$). The efficiency y of the source coding scheme is defined by

$$y = \frac{H}{v}$$

which is the ratio of the smallest average number of compressed digits per source letter that suffice for distortionless source reconstruction to the average number in the given distortionless coding scheme, where v is the average number of compressed digits per source letter.

Figure 7 shows the efficiency of UGSSC versus run length coding of length $n = 15$ with probability P of emitting a "one". Ancheta uses four block codes to achieve the compression which is shown in Figure 7:

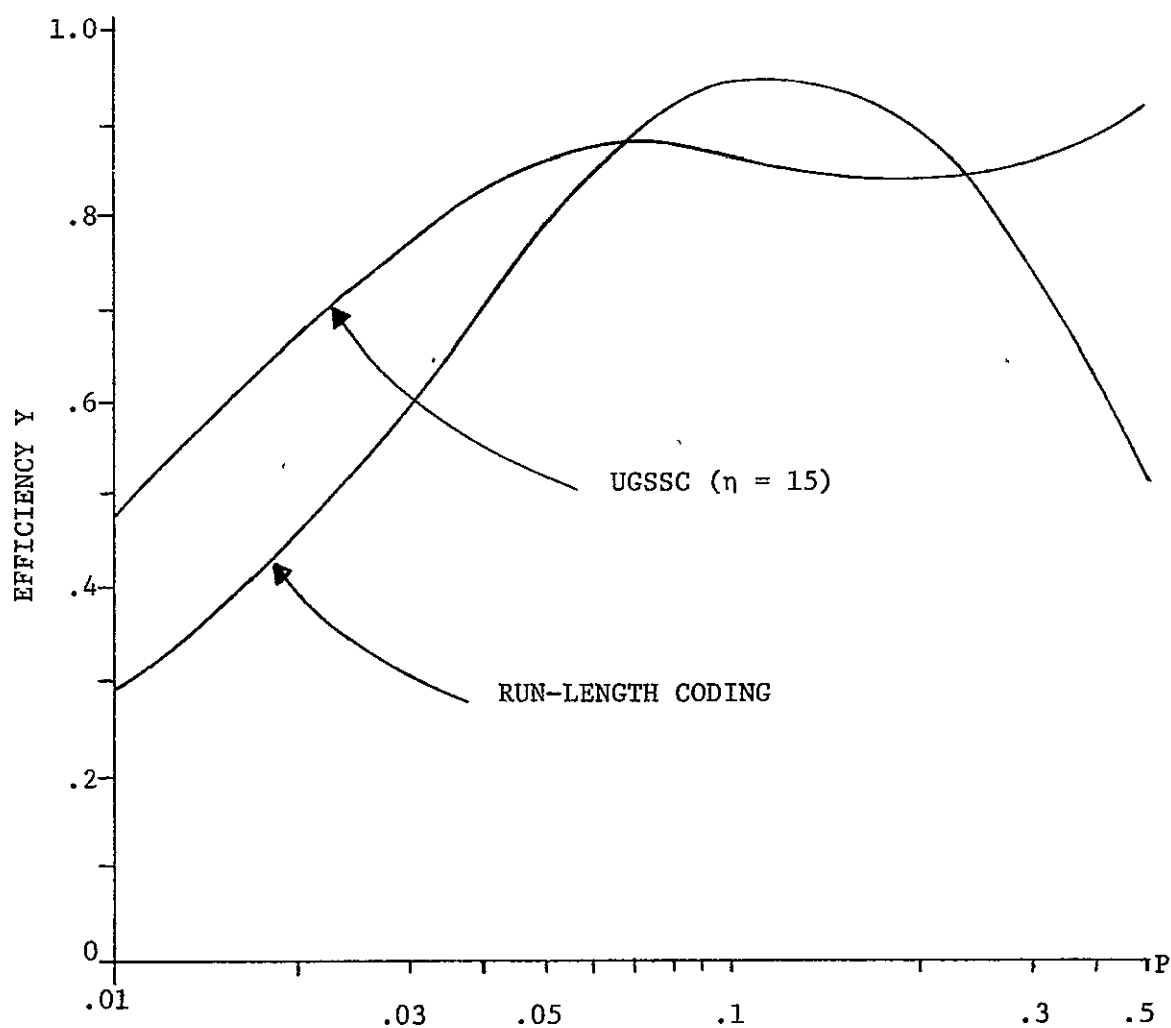


Figure 7. Efficiencies of UGSSC ($n = 15$), and run-length coding of length 15 requires for memoryless binary sources with probability P of emitting a "one".

1. The trivial (15,15) code so that S is the empty string,
2. The (15,11) Hamming code,
3. The (15,7) double-error correcting BCH code, and
4. A set of all zeroes $\{0\}$ the trivial code such that $S = e$.

But only syndrome former for the third code needs to be implemented. It is noted that error pattern estimators for the first and fourth code are trivial, for the third code is a simple threshold decoder [3], and for the second code is a simple combinational device.

ADAPTIVE SAMPLING

Adaptive sampling is a technique for adjusting the sampling rate of a given sensor to correspond to its information rate. The reasoning applied is that the data would not be redundant and therefore not compressible provided that the sampling rate was continuously and perfectly matched to the activity of the data source. Most of the time, present telemetry systems greatly over sample the data. For example, on a satellite, dc voltages that do not change for days or weeks are still sampled several times a second. In nearly all cases, the sampling rate is set on the basis of the fastest expected response for the source and not on the basis of the quiescent or normal value. However, to match the sampling rate to the data activity would require an activity detector for each data channel (incidentally, with extremely short response time and ideally with a predictor that varies the sampling rate just before the activity change). Another requirement is an extremely sophisticated programmer which will intermix a complex of sampling rates from many independently varying channels into a single data pulse train having a constant output rate so that the receiving station can synchronize with the

pulse train and thereby recover the data.

While, in principle, such a system makes the most effective use of bandwidth and keeps the sampling rate at a minimum, both on a pre-channel and a pre-link basis, it is extremely difficult to implement. This system will provide a data output whenever the change in data amplitude exceeds a predetermined tolerance on a per-channel basis.

REDUNDANCY REDUCTION

Redundancy reduction is a technique for eliminating data samples that can be implied by examination of preceeding or succeeding samples or by comparison with arbitrary reference patterns. Redundancy exists whenever the sampling rate of a multiplexer exceeds the frequency required to describe the input function in accordance with the accuracy requirements of the user.

The choice of reference patterns used to detect redundancy is virtually unlimited. Polynomials, exponentials, and sine waves are good examples of reference patterns by which real data can often be approximated. Arbitrary cyclic patterns, such as periodic components of an electrocardiogram or commercial television picture, can be used as a reference to detect redundant data from a given sensor. The process of redundancy reduction can be achieved by means of "prediction" from a priori knowledge of previous samples, or by a posteriori "interpolation" from future samples.

For redundancy reduction to achieve reasonable compression efficiencies, it is often necessary to introduce certain errors. These errors are caused by filtering and/or thresholding within the redundancy reduction process and do result in slight reductions

in the source entropies. However, unlike parameter extraction, redundancy reduction is designed such that the original source waveforms can be reconstructed with a guaranteed fidelity. This fidelity can be established to supply the data within the accuracy requirement of the user.

There are many possible techniques for redundancy reduction; however, those most effective and widely used are polynomial predictors and interpolators. Other mathematical forms, such as sine waves and exponentials, are more complex and generally not as efficient as the polynomials for most telemetry applications. There is a greater similarity between the polynomials and most real data.

Polynomial Methods of Redundancy Reduction

Predictors: A predictor is an algorithm that estimates the value of each new data sample based on past performance of the data. If the new value falls within the tolerance range about the estimated new value, it is rejected as redundant since it is known that the data value can be reconstructed within the specified tolerance using the previously transmitted data. Polynomial predictors are related to the family of linear predictors, which employs as its prediction criterion the assertion that the next data sample will lie on an n th-order polynomial, as defined by $n + 1$ previous samples. Thus, for this criterion,

$$y'_t = y_{t-1} + \Delta y_{t-1} + \Delta^2 y_{t-1} + \dots + \Delta^n y_{t-1} \quad (1)$$

where

$$n \geq 0$$

$$\begin{aligned}
y'_t &= \text{predicted data sample value at time } t \\
y_{t-1} &= \text{data sample value one sample period prior} \\
&\quad \text{to time } t \\
\Delta y_{t-1} &= y_{t-1} - y_{t-2} \\
\Delta^2 y_{t-1} &= \Delta y_{t-1} - y_{t-2} \\
&\quad \vdots \\
\Delta^n y_{t-1} &= \Delta^{n-1} y_{t-1} - \Delta^{n-1} y_{t-2}
\end{aligned}$$

Here, the $n = 1$ previous values, $y_{t-1}, y_{t-2}, y_{t-3} \dots y_{t-(n+1)}$ are known and y'_t is to be predicted. A basic type of telemetry data compressor is shown in Figure 8. The various implementations of this approach will be discussed.

Zero-Order Predictor Fixed Aperture

The simplest polynomial predictor is the zero-order predictor (ZOP) with $n=P$. In this case

$$y'_t = y_{t-1} \quad (2)$$

and the predicted value is merely the previous data point. A set of fixed tolerance bands are then set up as shown in Figure 9 with a width of $2K$. A sample value is transmitted only if it falls outside the aperture belonging to the last transmitted sample.

Zero-Order Predictor Floating Aperture

In the floating aperture algorithm of the zero-order predictor, Figure 10, an aperture of $2K$ is placed about the last transmitted data point. If y_0 was the last transmitted data sample, a prediction is made that subsequent data samples, $y_1, y_2, y_3, \dots y_5$ will be within k percent of y_0 . As shown in Figure 10, these samples are within the

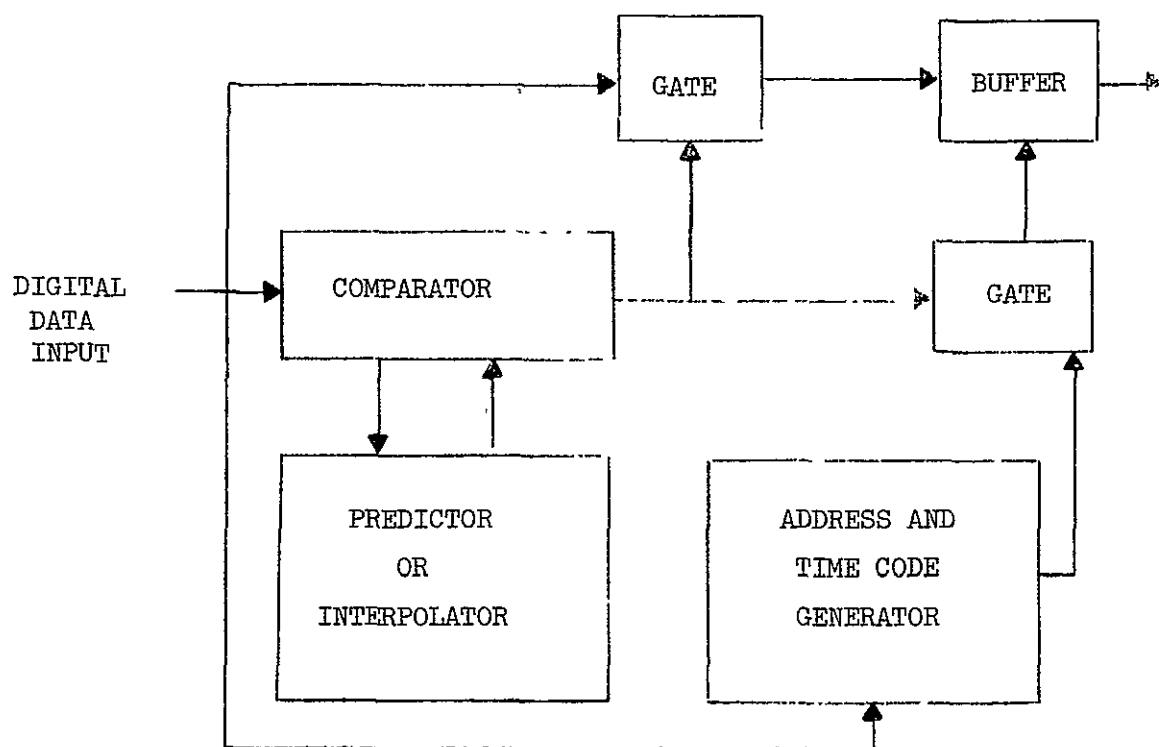


Figure 8. Basic Telemetry Data Compressor.

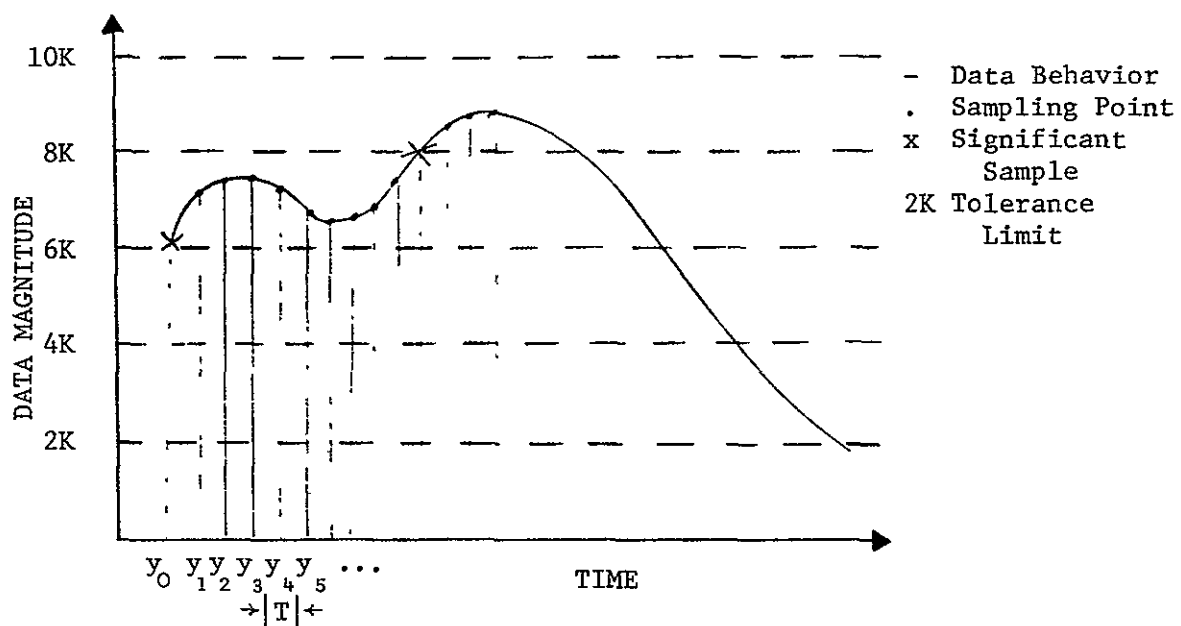


Figure 9. Fixed Aperture ZOP.

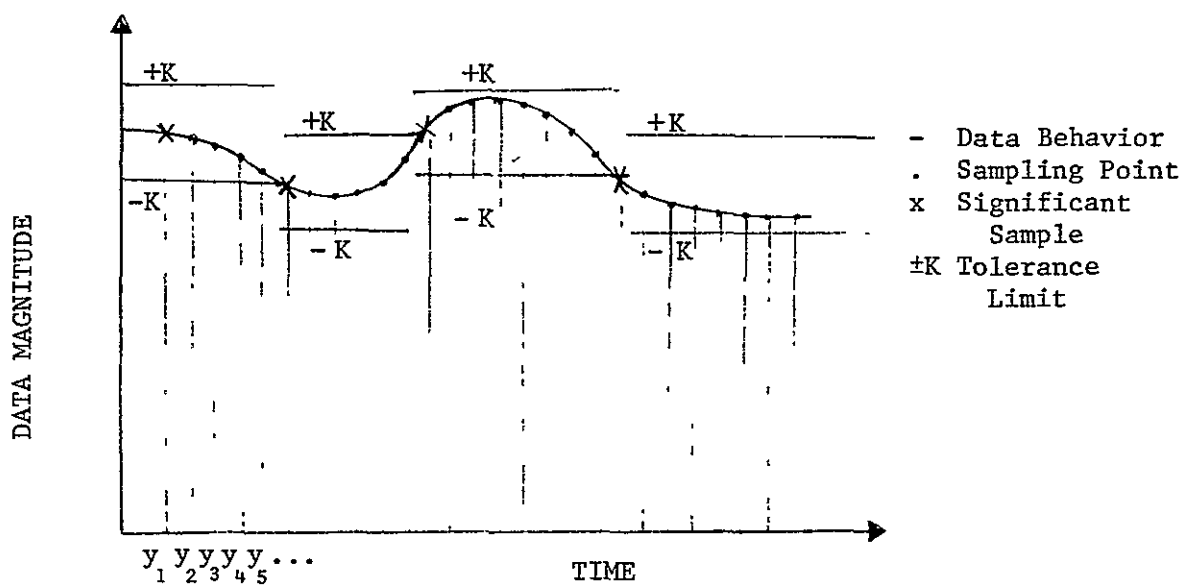


Figure 10. Floating Aperture ZOP.

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tolerance corridor denoted by $y_0 + K$ and $y_0 - K$ and can be discarded as being redundant. Each sample falling on or outside of the corridor must be transmitted as a nonredundant, or significant sample and is then used as the new reference for subsequent predictions. This use of the new, significant sample as the reference for future prediction results in a tolerance corridor which follows the data and is referred to as a "floating aperture." The term "zero-order polynomial prediction" implies that the redundant portion of a time function will be approximated by a horizontal straight line.

Zero-Order Offset Predictor

The zero-order offset predictor is a modification of the zero-order, floating aperture process, in an attempt to take advantage of knowledge of the data trend established at the time of the last transmitted sample. With this process, as with the previous technique, the predicted value remains constant as long as a sample is not transmitted. However, in this case the prediction is offset from the previously transmitted value by a fixed, predetermined amount. The sign of the offset is determined by noting the sign of the most recent out-of-tolerance deviation of the data. The offset would be in the positive direction if the deviation showed a positive trend and vice versa. The process is illustrated in Figure 11. The floating aperture of $2K$ is always centered on the predicted value. In Figure 11, the amount of the offset was arbitrarily chosen to be $0.75K$.

First-Order Predictor (FOP)

If in (1) the quantity n were set equal to one, we would have

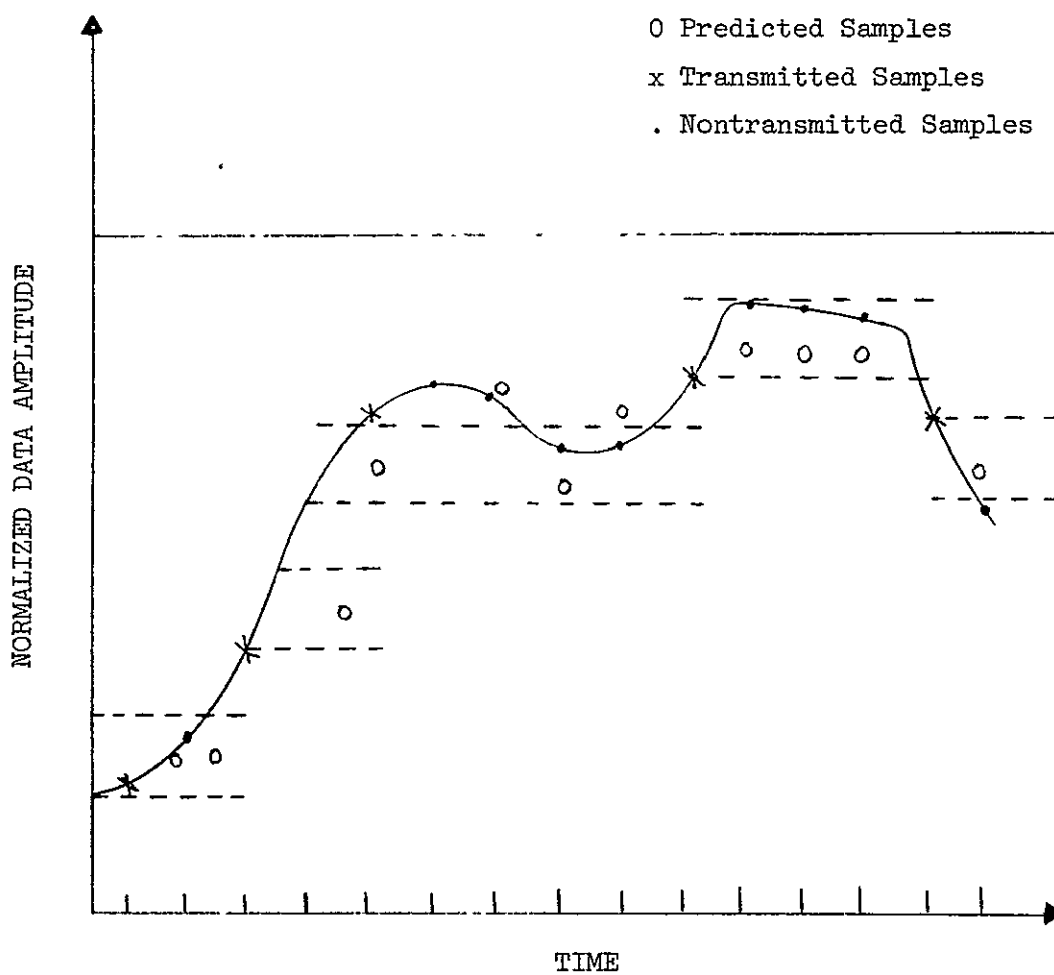


Figure 11. Data Sampling and Selection, Zero-order Offset Predictor.

$$\begin{aligned}
 y'_t &= y_{t-1} + \Delta y_{t-1} \\
 &= 2y_{t-1} - y_{t-2}
 \end{aligned}$$

In this case the prediction is made that y_t will equal the last sample value, plus the same change as the last value changed from the one before it. Gardenhire [7] has found a method of first-order prediction which maintains twice the maximum error as the original method but, under certain conditions, allows actual nontransmitted data samples to be used for prediction. In the original FOP it was pointed out that the predicted value of the data sample must be used to predict subsequent samples if that sample was not transmitted, in order to maintain a maximum error of K in sample reconstruction. However, in the latter, each within-tolerance data sample is held in storage until the out-of-tolerance test is made on the next data sample. When a sample falls out of tolerance the previous within-tolerance sample, which was being held in storage, is transmitted instead of the sample which fell out of tolerance. Thus, whenever an out-of-tolerance sample follows a within-tolerance sample, the prediction line will be defined by two actual data samples, rather than one actual and one predicted data sample as with the original FOP method. In this case for post transmission reconstruction the missing data samples must be reconstructed on the straight line joining each consecutive pair of transmitted samples in order to maintain a maximum sample value error of $2K$.

The polynomial prediction philosophy can be extended to include higher orders of polynomial redundancy reduction. Although higher-order polynomial predictors will, at times, provide a greater

compression efficiency on highly active data, experience has shown that for most telemetry data the simpler ZOP will provide suitable compression efficiency. If the power density spectrum of the data is known a priori, an optimum linear prediction technique by Schwarz [15] gives good compression results.

Interpolators

A prediction is a guess which to be effective requires that the characteristic of the data remain relatively constant from one time interval to the next. If the data are varying continuously in a random manner or if they are perturbed by high frequency noise, the redundancy reduction efficiency of the predictor will generally be low for reasonable system accuracies. Examining such data indicates that a greater number of redundant samples could have been eliminated if both future and past samples had been used. The process of after-the-fact polynomial curve fitting to eliminate redundant data samples is termed interpolation.

Zero-Order Polynomial Interpolation (ZOI)

The zero-order interpolation is similar to ZOP, Figure 12, in the sense that a horizontal line is used to represent the largest set of consecutive data samples within a prescribed peak-error tolerance. The primary difference pertains to the sample selected to represent the redundant set. The transmitted sample for the interpolator is determined at the end of the redundant set. Furthermore, the transmitted sample y'_t used for the interpolators is computed as the average between the most positive sample y_u and most negative sample y_e in the set. All samples in the set are within the prescribed peak error

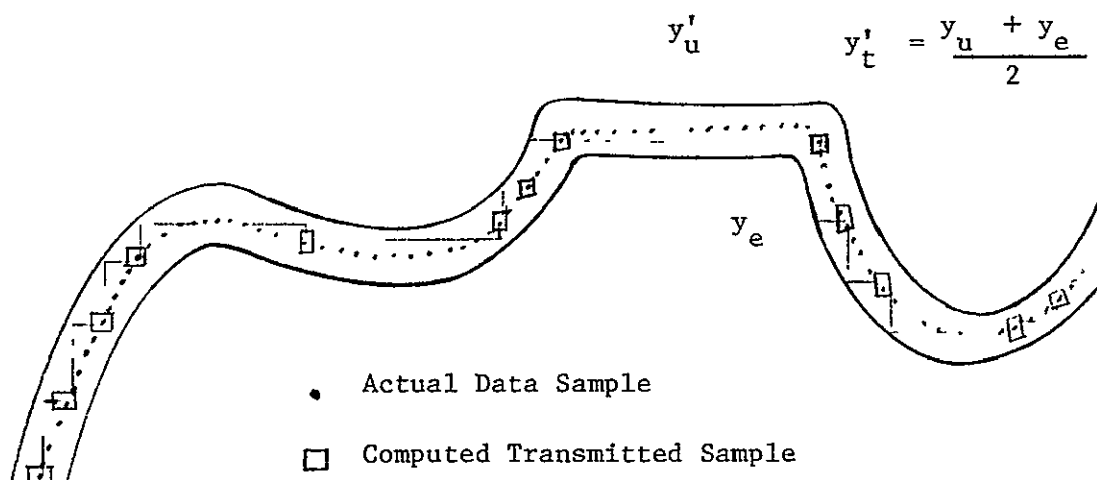


Figure 12. Zero-order Polynomial Interpolator.

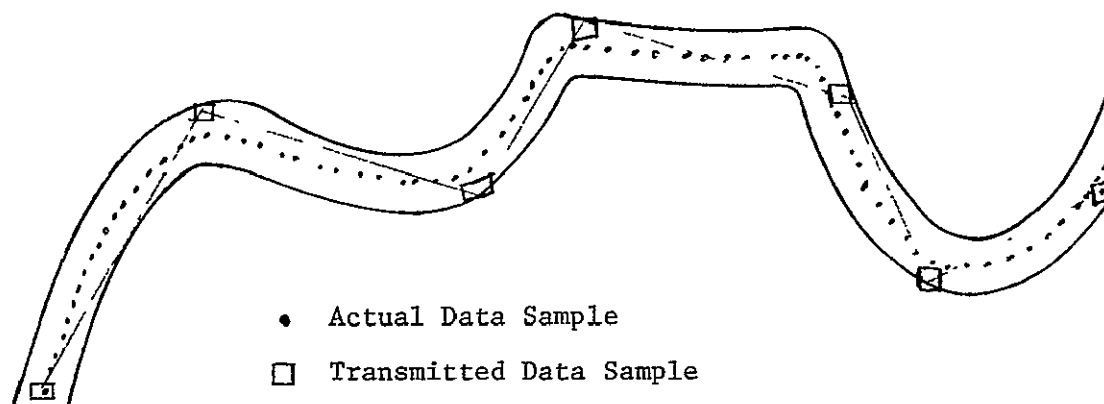


Figure 13. First-order Polynomial Interpolator.

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tolerance from the transmitted sample.

First-Order Polynomial Interpolar (FOI)

The FOI draws a straight line between the present sample and the last transmitted sample so that the intermediate data points are within a tolerance of the interpolated value on the straight line. In this algorithm, the first point is transmitted. A line is drawn between the transmitted point and the second sampled data value after the transmitted point. If the first point after the transmitted value is within a tolerance K of the interpolated value, then a straight line is drawn between the transmitted point and the third point after transmitted point. The interpolated values of the first and the second points are now checked to see if they are within a tolerance of the actual values. If at the K th sample value after the last transmitted sample value a line is drawn and the actual value differs from the interpolated value by a quantity greater than the tolerance, then the $(K-1)$ th sample is transmitted and the process is repeated. Figure 13 illustrates this technique and is sometimes called FOI-joint line segment or FOI--with 2 degree of freedom. To achieve the largest compression ratio, it is necessary to select a line segment that is within K percent of as many samples as possible. This optimum first-order algorithm requires freedom of both the starting and the end points of the straight line, resulting in four degrees of freedom. The performance of this optimum process is shown in Figure 14. It is seen that both the starting and end points of each line are computed values such that the length of each line is maximum for the peak-error guarantee. Also, the end point of one line segment can be connected

with a straight line to the beginning of the following line segment. Since the four-degrees-of-freedom FOI is a complex process to implement, it is reasonable to use the first technique, which can be mechanized more easily [12] and [13].

Implementation and Analysis

Figure 15 illustrates a typical data compression system structure with subsystem details in Figures 16 and 17. Figure 18 illustrates the typical decompression system. These figures should be reviewed while reading this section.

At this point it is possible to define one measure of data compression efficiency which has been termed "element compression ration." This is the ratio of the number of data values presented at the input to the number of significant data values delivered to the buffer memory during a specific time interval. While this is a basic measure of ability of a particular algorithm to remove redundancy, it is not the significant measure of the efficiency of a data compression system to conserve bandwidth and/or power. The difference arises from the necessity to identify the significant data values with an address, a location, and/or time of occurrence so that it will be possible to reconstruct a time history for each data source within prescribed error tolerances. It is also necessary to provide synchronization information in the output bit stream. Therefore, a second measure of data compression efficiency has been defined as "bandwidth compression ratio." This is the ratio of the number of bits presented at the input to the number of bits delivered at the output of the data compressor. This ratio includes all penalties for identification,

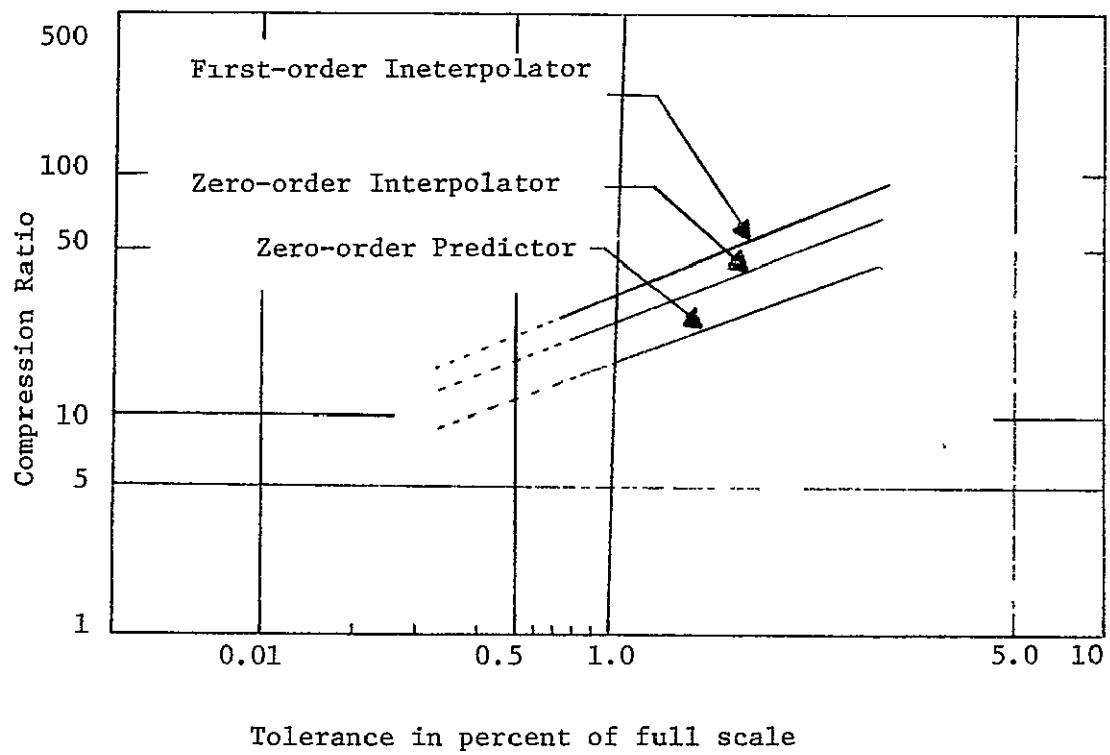


Figure 14. Compression Characteristics of Different Techniques.

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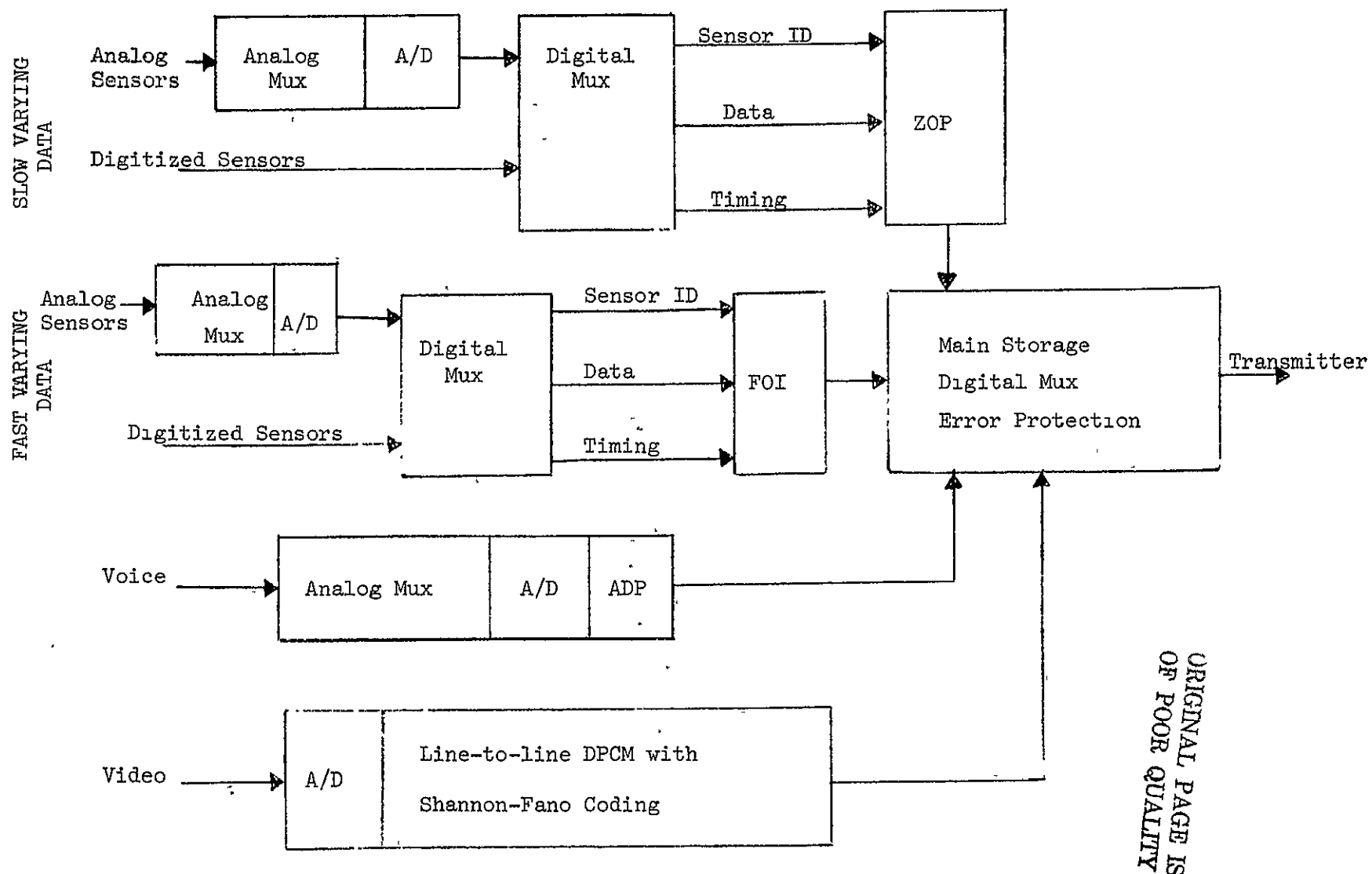


Figure 15. Typical Compression System

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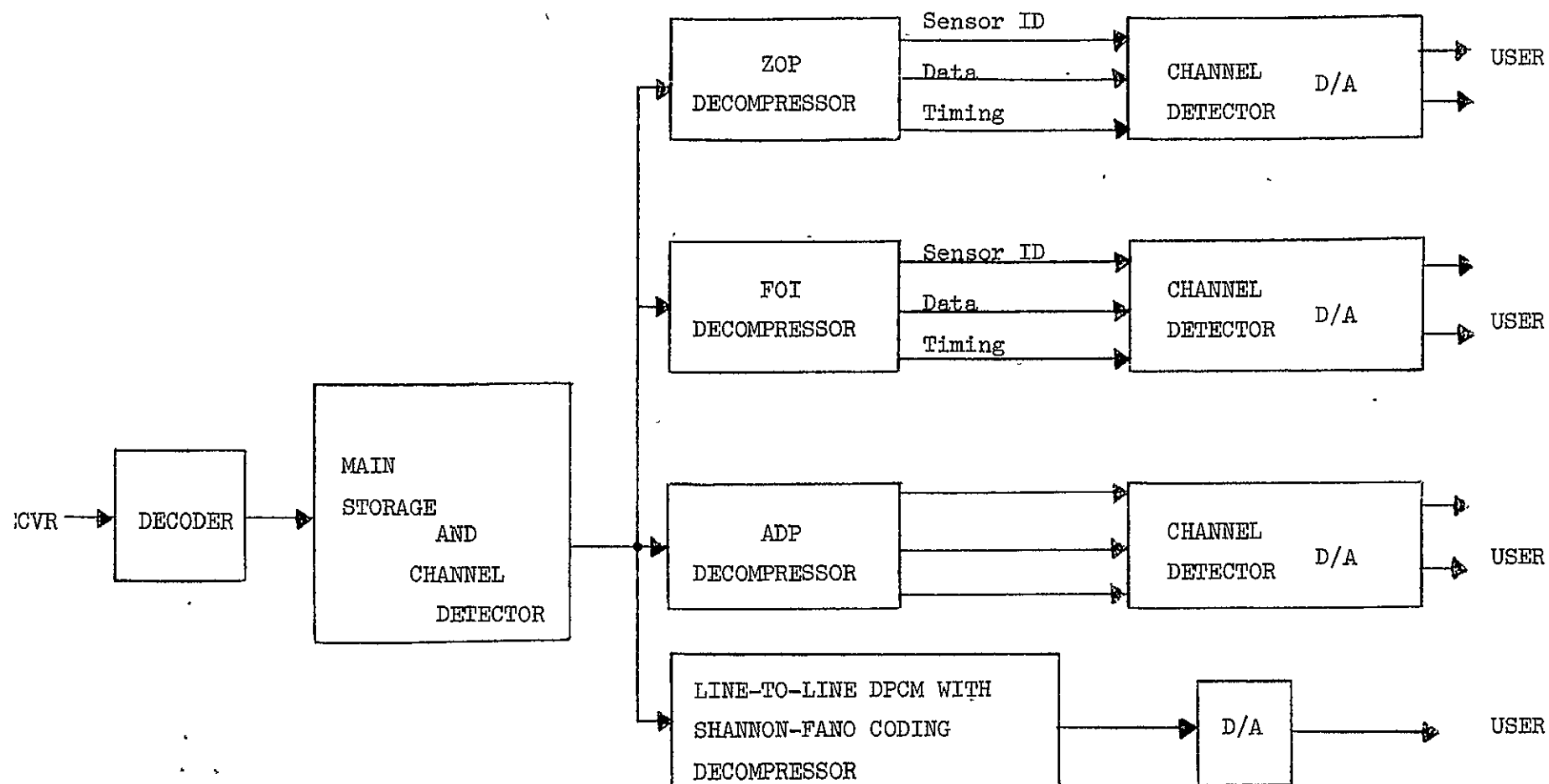


Figure 16. Typical Decompression System

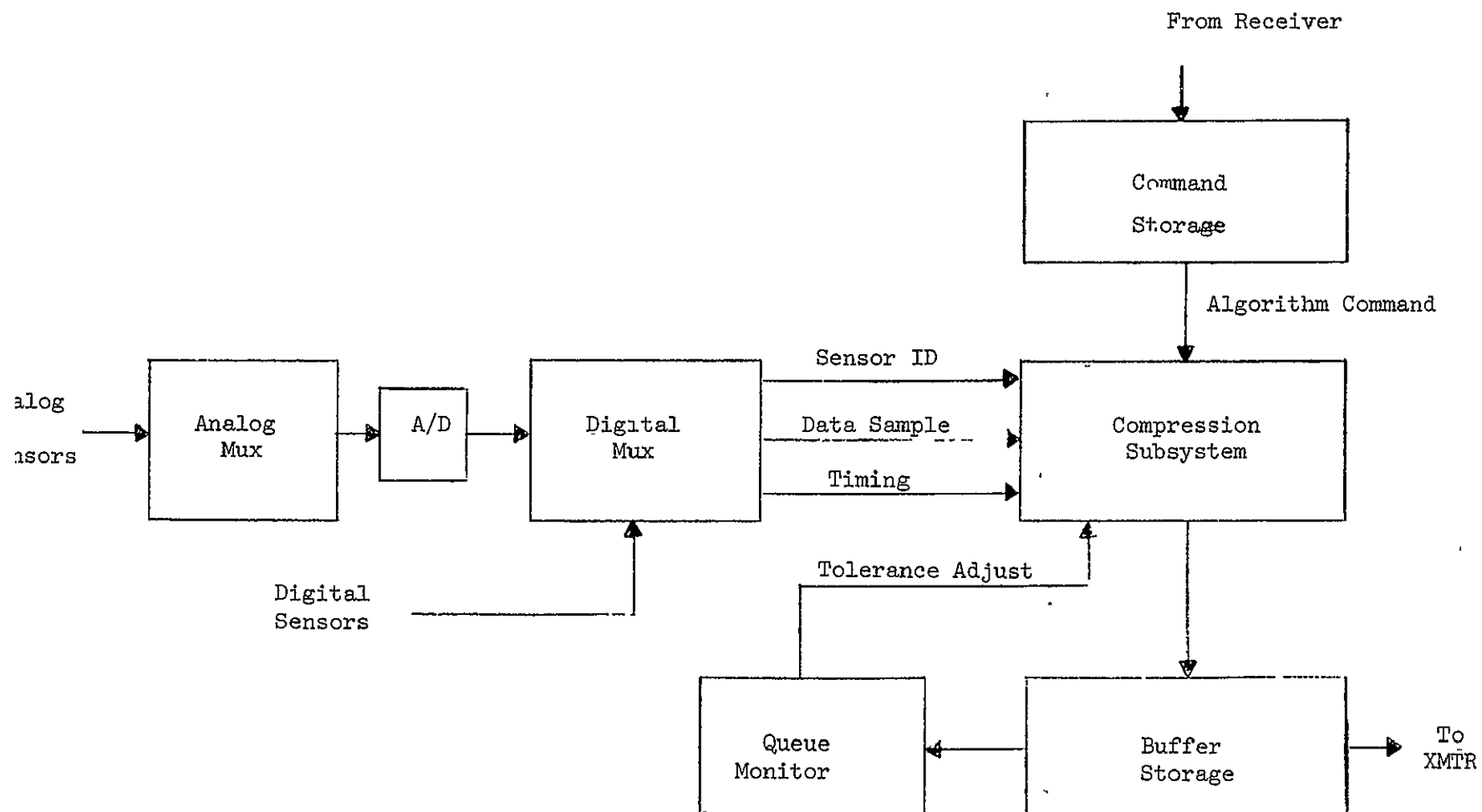


Figure 17. Typical Data Subsystem

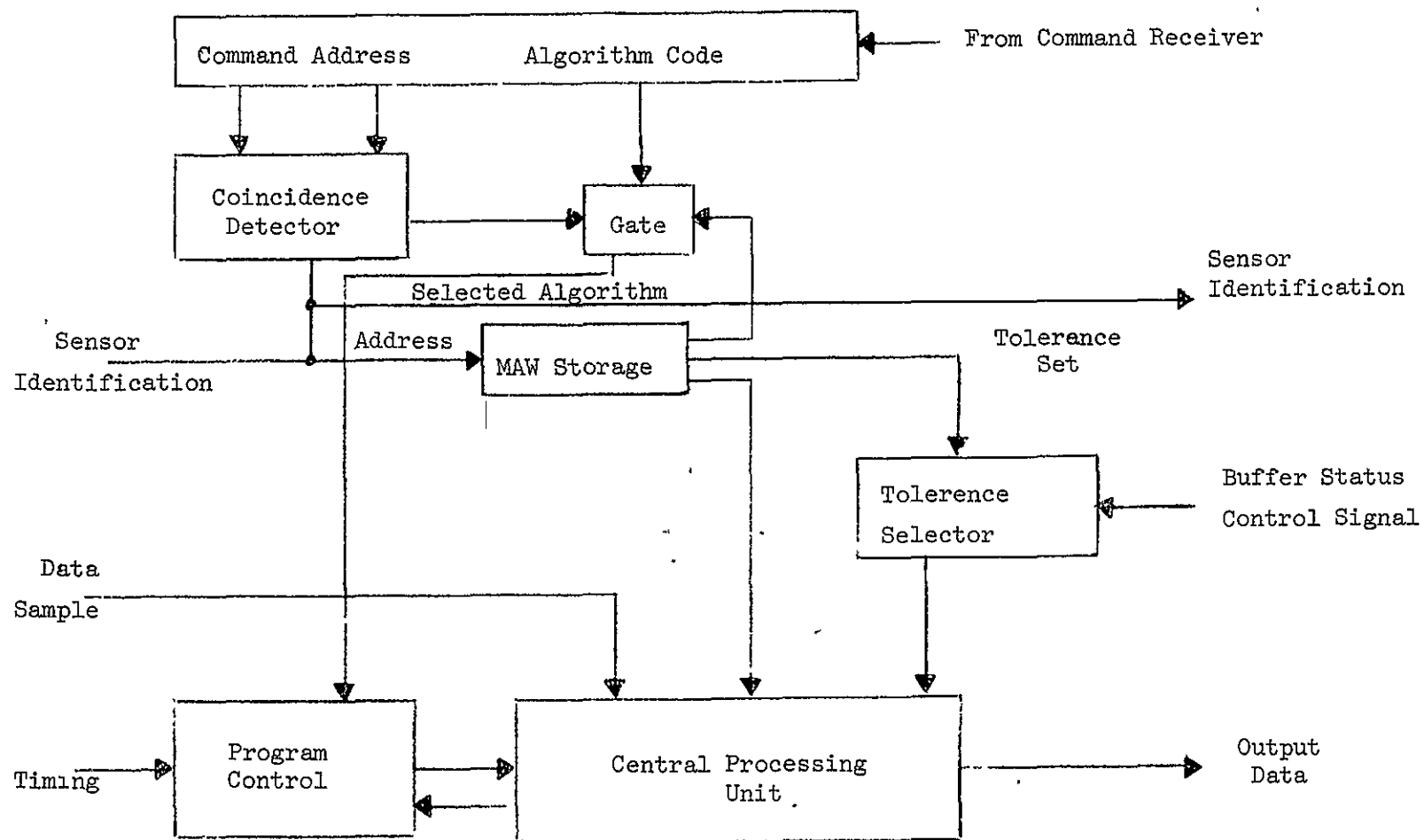


Figure 18. Typical Compression Subsystem

timing, and synchronization and is therefore a true measure of overall compression efficiency.

The operation of the data compressor is controlled by the timing and control logic. The first of these is buffer loan or fullness control. Design of the output buffer is one of the most important tasks to be faced in implementing a data compression system. Upon proper design of the buffer, including such parameters as size, input-output data rates, and occupancy control, rest the over all compression efficiency and error performance of the system.

The buffer permits the efficient merging of nonredundant samples from several sensor channels into one constant-rate data stream for transmission. There is a need for buffer load control, because of the possibility that, over a period of time, the number of significant data values read into the buffer will exceed the number read out over the same time period. If the buffer is large enough, all the data can be stored and eventually read out, provided that the average input rate drops below the read out rate for a sufficient period. However, in most practical applications, unlimited buffer capacity is not available and therefore there is a danger of buffer overflow during period of high data activity. If the buffer is permitted to overflow, there is an uncontrolled loss of data and no way even to approximate what has happened. Several methods of preventing overflow have been developed. Among these are the deletion of certain low-priority data to enable handling greater activity in high-priority data. Another method is to increase the tolerance band on all or a selected number of data sources to reduce the number of data values accepted as significant. All methods result in degradation of some or all of the

data, but in a controlled manner so that it is still useful. Buffer consideration has been discussed by Schwartz [15].

Another function that can be realized is effective presampling filtering, accomplished by utilizing the data compressor memory and some additional computation and logic capability. It is considered good design to filter all input signals prior to multiplexing to avoid frequency foldover of unwanted instrumentation noise (commonly called "Aliasing errors"). However most instrumentation systems cannot afford the luxury of presampling filtering because of size, weight and cost considerations.

Adaptive precompression filtering has many attractive features as a means for controlling the queue length of the buffer, [8]. The philosophy behind adaptive buffer control is that it is better to sacrifice selected data or data accuracy in performance to an arbitrary loss of significant data caused by buffer overflow. Adaptive aperture control reduces the accuracy of the data regardless of frequency. On the other hand adaptive filtering reduces the accuracy of only the higher frequency components in those data having high activity (the ones that are causing the buffer problem). However, the extent of the usefulness of this technique has not been fully established.

Another problem arises during periods of normal data activity, when it is expected that the buffer will tend to be empty. To prevent buffer underflow, redundant samples may be transmitted. If these redundant samples are taken from data channels from which no recent sample has been transmitted, because of inactivity in those channels, additional confidence in the received data can be provided. Feedback for controlling buffer occupancy is derived by the fullness-level

detector.

Specifications for data compressors employed in space vehicle telemetry systems must be determined largely from knowledge of previous data and from anticipation of future data behavior. However, the exploratory nature of space vehicles often produces data of unexpected behavior. To cope with this and other possibilities, it is desirable to have provisions to modify the data compressor performance in flight.

Decompression of Data

Another factor to consider is that of "decompressing" the data; that is the receiving end of the link, either restoring the data to its original form or putting it into suitable form for its end use. Most of the work in the area of telemetry has been done during the sixties, and a majority of the papers written in this period. Most papers deal entirely with the compression part of the problem; However, there are some basic differences in the algorithm that affect the reconstruction problem. As defined in this paper, the predictors are relatively easy to decompress. Since decompression is the inverse of compression, redundancy must be reinserted into the original. For analog decompression this is easily done by using a uniform time base such as may be provided by a strip chart recorder. The indicator (pen, styles, etc) is set to the initial value and draws a straight line with the predicted slope (a horizontal line for ZOP) until a new value is received. This means that the original data had varied beyond the bounds of its tolerance band and therefore the reconstructor must shift to the new value and continue until another value is reached. This method will provide sharp-cornered approximation of the original signal data which

may be smoothed by various methods. However, the interpolators as defined herein present a very different problem for receiving and decompression. For these algorithms the representative data values and slopes are not known and therefore, not transmitted until a longest possible line segment has been fitted to the data. This makes it impossible to plot a reconstructed replica of the original data without imposing some delay. How much delay becomes a problem.

There is no clearly superior method for all sources under all conditions; with this in mind, Table III gives a comparison between different techniques mentioned herein for PCM Telemetry, where

FOI-2DF = FOI-two degrees of freedom

FOI-4DF = FOI-four degrees of freedom.

Using multiplexers we can use one algorithm for different data channels (i.e., designing ZOI in general algorithm and with a central processing unit (microprocessor) control the tolerance to the accuracy desired for each channel ($\pm K$ percentage of full scale). By affording a larger buffer, controlling the buffer fullness and having preprogrammed the microprocessor for channel priority, also, during normal operation it is probable that the buffer will become empty. Having preprogrammed the microprocessor, at these times we would sample channels that have not been active for some time (specifically for those for which we use interpolators). This keeps the reliability high and results in a more efficient overall system performance.

Table 111

Summary of Trade-off Factors

Algorithm	Performance Based on PCM Telemetry	Implementation Effort Required	Impact on Data System
ZOP	Good on all data types Especially appropriate for slow varying data (i.e., ≤ 1000 Hz variation rate in data values). Data reduction varies as accuracy desired [10:1 to 40:1]	Moderate	small-ground reconstruction is straightforward
FOI-2DF	Good, particularly for large aperture; not suitable for slow varying data. Good for fast varying data (i.e., ≤ 1000 Hz variation rate. Data reduction varies as accuracy desired [15:1 to 80:1]	Moderate	Small
FOI-4DF	Good on all data 5% more compression than FOI-2DF	Requires large and variable number of calculation	Small
Linear-predictive Differential Coding (LPDC)	\approx zero errors data reduction (20:1)	coding implementation is difficult	Variable word length recovery problem

In comparison to Table 1 of Chapter 11; note that for voice and telemetry data all the techniques of this table perform better than PCM and DPCM and ADM, but performance of APD and LPE are comparable with these techniques when between 0.75 to 0.9 percent of full scale tolerance is acceptable, and with about 1% of full scale tolerance are comparable with vocoders. LPDC performance is better but the trade off is in the complexity of the recovery.

With the size of memories* getting smaller and the price getting cheaper it is more practical to use microprocessors to control the data compression systems.

*Magnetic-bubble memories being commercially available [63] at low cost (\$75 for 92 Kb memory whose transfer rate is 44 Kb/s, average access time 4 ms, power dissipation 11.5 W, weight 0.69 lb, and volume 38 in³). These are relatively cheap for small size usage (micro-processor applications), but double the density and twice the access time speed is expected by 1980. When large storage with faster access time is to be used, the beam addressed metal-oxide-semiconductor (EAMOS) are attractive at 32 Mb to 600Mb or more [51], access time 30 μ s and 10 Mb/s data transfer rate at moderate price results in a good system efficiency.

CHAPTER IV

IMAGE CODING TECHNIQUES

Introduction

The function of an image transmission system is to convey to an observer a "best" reproduction of a distinct scene or document, subject to limitations which may be imposed by the cost and/or technical characteristics of the channel and terminal equipment. The original scene may be three-dimensional, colored, and moving, and is normally viewed binocularly. Hence, it is clear that a monochrome, monocular, sampled, and quantized representation is at best a very crude approximation. It is also possible that what constitutes the "best" representation may vary according to the application. In image data compression one is concerned with converting an analog picture, often generated by optical sensors, to the smallest set of binary integers such that this set of binary digits can be used to reconstruct a replica of the original image.

One may classify compression techniques as pure statistical, psychophysical, or a combination. One of the most important areas of image coding is psychovisual properties of human vision and related coding techniques. Its importance is due to the fact that still nearly all images are viewed and evaluated by human observers, and having a distortion measure that would have strong correlation with the perceived quality of images is of paramount importance. A human observer seems to process the image by a bank of parallel mechanisms, each of which are sensitive to different regions of the two dimensional

spatial frequency plane. The mechanisms are sensitive to contrast and not absolute intensity and are much more sensitive to large errors than small; and their sensitivity is reduced in those regions where the original image has large changes in contrast, also it takes a substantial fraction of a second for observer to recover their spatial acuity after a scene change. Surely if part of this wasted channel capacity were applied to some other aspect of the original scene, the overall effect would be better.

A word of caution concerning the assessment of the performance of different encoding techniques. At the very minimum, such assessment requires the measurement of the bit-rate for a given picture quality. It is primarily the assessment of picture quality that is so variable. Picture quality depends on many factors, for example lighting conditions and monitor adjustments, the range of picture material that is presented, whether a single stored frame (or photograph) is being viewed, the amount and type of movement contained in the scene and the experience of the viewers. The difficulty in comparing picture quality, in turn, complicates the task of comparing different coding schemes. Even side-by-side comparison of picture quality would not provide definitive rating since other factors such as error performance, cost, complexity and compatibility will all effect the final decision on the type of coder most suitable for a given application; with this in mind we look at different image coding schemes.

Interframe Coding

A moving picture can be represented as a set of frames of still

pictures, where each frame corresponds to a certain instant in time. The frames are presented at frequent intervals of time so that the eye can perceive the picture without distortions. In accordance with the standards of the National Television System Commission (NTSC), the picture is presented 30 frames a second in U. S. commercial TV broadcasts. Since the picture is presented at this high frame rate, there is considerable correlation between successive frames of the picture elements within a given set of frames. Hence a three-dimensional transformation of a block is used to reduce spatial and temporal correlations of the picture elements within the block. Roese [56] has shown that this approach yields transform coders whose performance greatly exceeds that of conventional interframe coders. However the main disadvantage of such coders is that they may require excessive storage, as in the case study reported in [56] where the data are processed in $(16 \times 16 \times 16)$ blocks. Natarajan-Ahmed [73] are considering much smaller block sizes, namely $(4 \times 4 \times 4)$. As such, storage requirement are substantially reduced and may be a level which will be cost effective in 1980's due to the fact that rapid advances are being made in charge couple devices (CCD) memory technology.

It is apparent that interframe coding via three-dimensional transforms involves time consuming address evaluations related to three dimensional arrays. Natarajan-Ahmed [73] has shown that a Kronecker product formulation leads to an efficient algorithm that enables the vector processing of three-dimensional arrays. Their experimental results are given, using the luminance component of (NTSC) video signal sampled at approximately 8 mb/s, which corresponds

to 512 picture elements per line. There are 525 lines in a frame, and each sample is quantized using six bits. The picture element in four frames of a picture were read into computer and divided into $(4 \times 4 \times 4)$ blocks. Considering only pels (pel \equiv 1 picture element) that correspond to the visible portion of the video signal; i.e., approximately 416 pels in horizontal direction and 476 pels in vertical direction. The picture elements within each $(4 \times 4 \times 4)$ block were arranged in lexicographic order to form (64×1) dot vector and were processed and reconstructed; the author claims a 6:1 compression at 1 bit/pel.

Roose, Pratt, and Robinson [65] have made an extensive set of computer simulation studies to experimentally evaluate the performance of interframe transform coders. The hybrid coders considered are parametric in many variables including choice of separable transforms, zonal sampling or zonal coding with various quantization strategies, presence or absence of channel error, and average pixel bit rate. One sequence contains images from a fixed position camera of a moving subject engaged in conversation. The others contain images of a plant photographed from an airplane in a flyby trajectory. Each frame is digitized at a spatial resolution of 256×256 pixels with each pixel amplitude linearly quantized to 256 levels. Visually, no image degradation can be seen for bit rates as low as 0.5 bits/pixel/frame for three-dimensional cosine transform coder; beyond this value some blotchiness becomes apparent.

Performance of adaptive hybrid transform/DPCM coder in the presence of noise has been investigated by computer simulation of binary symmetric channel transmission system, at average pixel bit

rates from 0.1 to 1.0 bits/pixed/frame with 16 x 16 subblocks. The results indicated that, even at the lowest bit rate stability of coder performance is achieved within the first eight frames. Also, coding results obtained with $p = 10^{-4}$ (bit error probability) are essentially indistinguishable from the case of $p = 0$. Results also indicate that the performance of hybrid interframe coders are dependent on the temporal correlation, reduced levels of performance are to be anticipated for image sequences distorted by effects of camera motion. Compression ratio of 8:1 for fixed camera and 4:1 for moving camera is obtained.

R. G. Gaskell, P. L. Gordon [67] have done research in interframe coding of 525-line monochrome television for possible application to video conferencing or video telephone.

This method is based on the assumption that the scenes consist of stationary background areas with moving objects in the foreground, and TV cameras will not be moved very often. With conditional replenishment [18] most of the transmitted information is devoted to moving objects in the picture, i.e., those areas not already at the receiver. Moving areas are detected by a segmenter which examines significant frame-to-frame differences, eliminates as best it can the one's due to noise, and blocks in small gaps between differences in order to ease the addressing requirement. Once defined the moving-area pels are efficiently coded using higher order frame-toframe DPCM and passed to elastic storage (buffer) to await transmission by the channel to the receiver buffer. At the decoder the received pels are inserted into the appropriate locations in a frame memory and the resulting picture is displayed.

During periods of rapid movement too much data are generated to be sent right away and buffer overflow is a problem which could be overcome by means such as those discussed in the previous chapter. This algorithm provides good quality pictures at 1 → 3MB/s for the conditions mentioned above (.19 bits/pel), the main disadvantage of this technique is the same as in other interframe coding techniques; use of extensive memory storage. However, this may not be a significant problem in the next decade. Table IV summarizes the above discussions.

Three-Dimensional Spatial Non-Linear Predictor

Efficient encoding systems generally fall into two categories, predictive coding system like DPCM [9] and transform coding [37]. However, a DPCM system as well as transform coding, must be matched to the statistics of the image being encoded. When there is much redundancy to be removed, the performance of these coders may be sharply degraded with images whose statistics are markedly different from those for which the coders are designed.

The non-linear prediction systems studied so far use intra-frame switched predictors which based their prediction on "previous" pels in the neighborhood of the pel to be predicted. These systems had the disadvantage that an error in the predictor input would propagate throughout the image. The three-dimensional spatial non-linear predictor by Dukhovich and O'Neal [84] from those previously proposed by forming a prediction as weighted sum of three non-linear estimates called representatives. Each representative is a non-linear estimate based on the pels in a plane containing the pel to be predicted. An

Table IV. Comparison of Different Techniques for Monochromatic Image Encoding

Technique	Scene Variations	Performance	Implementation
7 bit PCM 325 lines 4 MHz monochrome sampled at twice the Nyquist rate	Normal conditions	56 Mb/s - Good results	Simple to implement
Interframe DPCM at 3 bits/pel	Fast varying signal	24 Mb/s - Good results	Reasonable complexity plus storage problems
Interframe Transform Coding at 2 bits/pel [37]	Slowly varying scenes	16 Mb/s - Reasonable Results	Reasonable complexity plus large storage problems
Interframe Transform Coding can be done at 1 bit/pel	Moderately varying scene	8 Mb/s - Reasonable results	Complex plus large storage requirement
Interframe DPCM pulse conditional replenishment coding at .75 bits/pel	video-conference	6 Mb/s - Reasonable results	Complex plus large storage requirement
DPCM Picture Coding with Adaptive Prediction [75]	Fast varying scene	34 Mb/s - reasonable	Simple to implement
Adaptive Delta Modulation [70]	Moderate movement	1 → 3 bits/pixed	Simplest of all to implement

error in the input of such a predictor decays fairly rapidly and does not result in catastrophic failure of the predictor.

To verify the later, propagation of error, the system was simulated and its performance was compared with linear predictor. The result is that linear predictor performs as well as the non-linear predictor in the volumes with low motion and large details. However, for the volumes with small details or rapid motion, non-linear prediction is significantly better than linear prediction.

The Color Signal as a Source

In laying the foundations of present day color television standards in the late 1940's, much study went into various background topics such as colorimetry and visual preception so as to match the resulting signal to the color fidelity requirements of the human observer. Additional studies have been made in areas of threshold color difference preception and on the interaction between the "brightness" (luminance) and "color" (chrominance) component of the signal.

From the first experiments in color encoding, two somewhat separate approaches have been exploited. The first is to operate directly on the composite television signal, whereas the second divides the signal into three components, codes each component separately and then, after transmission, combines them again to form a composite signal.

Color Representation

The foundation of 3-color colorimetry lies in a series of rules generally attributed to Grassman [1]. Two of the more important rules can be expressed as follows:

$$(c) = \alpha (P_1) + \beta (P_2) + \gamma (P_3)$$

where (c) is color that is to be matched by α units of color (P_1), β units of color (P_2) and γ units of color (P_3). The colors (P_1), (P_2), (P_3) are conventionally called primaries.

2. The luminance* of a color is equal to the sum of the luminance of the components in the mixture;

$$L = L_1 + L_2 + L_3$$

Instead of referring directly to the number of units required to make the match, colorimetrists use normalized quantities called chromaticity coordinates expressed by the relations:

$$r = R/(R + G + B)$$

$$g = G/(R + G + B)$$

$$b = B/(R + G + B)$$

Here we have changed the notation such that the units α , β , and γ , which are called tristimulus values, have been replaced by the symbols R, G, B, respectively, to agree with the common usage of Red, Green, and Blue primaries in these experiments. In a three-primary match to the reference white, the ratio of each of three component luminances contributed by each primary to the total luminance are called the luminosity coefficients.

The Commission International de L'Eclairage (CIE) in 1931 defined the color-matching data of the standard observer to be the mean of the

*Luminance is a quantity measured by a photometer. It is the photometric brightness of a uniform, small field.

mixture, on the other hand tristimulus values express how much of a primary is needed in a match to a given spectral color.

Color Television Signals

From the colorimetry view point we can draw an analog between a color television system and a colorimeter. The three phosphors of the receiver correspond to the three primaries of colorimeter, and the camera taking filters correspond to the color mixture covers for these primaries. Each color is the same before the camera and must be matched by suitable controlling of the light output of the receiver phosphors. This goal will be achieved if the light contribution from the receiver phosphors are adjusted to be equal to the tristimulus values appropriate to this system of primaries for each of the colors in the original scene.

The National Television Standards System Committee (NTSC) format of color television signal is achieved by transmitting a luminance signal representing the monochrome information;

$$E_Y = 0.30E_R + 0.59 E_G + 0.11 E_B \quad .$$

The coefficients in this equation sum to unity so that when, E_R , E_G , E_B were adjusted to be equal on the reference white, coefficients in the above equation sum to unity so that when Illuminant C is reproduced:

$$E_Y = E_R = E_G = E_B$$

another signal is also transmitted comprising two "chrominance" components which supply the additional information needed to represent the chromacity of the original scene. The NTSC color difference

signals denoted as I and Q were chosen to be:

$$E_I = ((E_R - E_Y) \cos 33^\circ / 1.14) - ((E_B - E_Y) \sin 33^\circ / 2.03)$$

$$E_Q = ((E_R - E_Y) \sin 33^\circ / 1.14) - ((E_B - E_Y) \cos 33^\circ / 2.03)$$

This choice is near optimum for the constraints involved. The E_I signal carries orange-red to cyan information and E_Q signal carries green to magenta information.

The Phase Alternation Line (PAL) color television system is in use in most of Western Europe, and the SECAM* system is in use in France, Iran, the Middle East and Eastern Europe [19]. In PAL and SECAM, the chrominance components are designated as

$$U = (B - Y) / 2.03$$

$$V = (R - Y) / 1.14$$

in SECAM, the luminance signal is transmitted on every line, and a delay line in the receiver allows repetition of each chrominance signal so that Y, U, V signals are simultaneously available.

A word of caution concerning the assessment of the performance of different encoding techniques. At the very minimum, such assessment requires the measurement of the bit-rate for a given picture quality. It is primarily the assessment of the picture quality that is so variable. Picture quality depends on many factors, for example, lighting conditions and monitor adjustments, the range of picture material that is presented, whether a single stored frame

*The name SECAM is not an acronym. It refers to the use of a sequential chrominance signal and a memory device (sequential a memoire).

(or photograph) is being viewed, the amount and type of movement contained in the scene and experience and expectations of the viewers, which brings about the psychophysics of color vision, and the psychological aspect of image coding discussed in the beginning of this chapter; plus, in color television, masking effects are concerned which pertain to the influence of the signal at the point. In terms of the components of a color television signal, our interest will be restricted to the effect of masking of chrominance signal by the luminance signal, and the masking of the chrominance signal by the chrominance signal. For more detail see Judd [11].

Signal Statistics

Starting with an RGB signal, we will assume that each component has a range 0 - 1.0 unit such that equal quantities of R, G, and B produce an achromatic mixture. The probability density function of the R, G, and B signals varies from picture to picture, but one could postulate that if an average were taken of a large number of pictures, the resulting probability density would be much more uniform, just as one experiences with the luminance signal.

From this point on we will assume we have a luminance signal and two chrominance signals and study the properties of just the chrominance components.

A television system can only reproduce chromaticities falling within the triangle formed by connecting the chromaticities of the three display - tube phosphors [2]. Rubinstein [40] and Pirsch [60] have used joint entropies to study the statistical dependence between the components of a color signal. Measurements of cross-correlation

coefficients between components would be an alternative method of measuring statistical dependence, but using an entropy measure, a quantity more directly related to bit-rate reduction is obtained. For use as test pictures the average value of 11.22 bits/joint-sample was obtained for the quantity $H_s = H(Y) + H(U) + H(V)$ where Y, U, V were individually quantized with an accuracy of 6 bits. Conclusions that can be drawn from these studies is that redundancy between the amplitudes of the differential signals of the components of a well chosen color signal, although present, is not very significant, and if exploited will not lead to any large reduction in bit-rate.

Reducing Color Image to PCM

If R, G, and B signals are to be digitized, it is not necessary to quantize each with the same precision since they are not equally sensitive to added noise [5], for example, for a picture only containing the three primary colors displayed at a constant luminance, the threshold signal-to-noise ratio of 36 dB for blue, 41 dB for red and 43 dB for green, although these figures vary somewhat with luminance. The drawback of coding the components of RGB signal is that relatively high spatial resolution is required by each component. In Der Smitten [50], this problem of increased RGB requirements is partially overcome by the splitting of the input into a lowband (0 - 1.25 MHz) RGB signal and a highband (1.25 - 3.75 MHz) luminance signal. The RGB space was quantized three-dimensionally into 125 volumes. The boundaries of these volumes were determined by subjective experiment. The RGB signal requires 7 bits at a sample rate of 2.5 MHz, while the highband signal would require perhaps 3-bit DPCM

at 7.5 MHz, giving $17.5 - 22.5 = 40$ Mb/s. The RGB signal requires a relatively large fraction of the total bit-rate, and with just 125 colors some color-containing is visible in coded pictures. In practice the television signals are gamma corrected prior to transmission, and this nonlinear operation significantly affects the quantization process [45].

Transmission errors on a PCM coded signal is [39] to shift the display luminance slightly toward a mid-value. Similarly, chromaticity values are shifted towards the average value of the coded signal which is the white point for RGB and YIQ. Transmission errors appear to have least effects on the RGB signal which is perhaps expected, since these signal components have the most uniform distribution of energy.

DPCM

The overall efficiency of coding technique will depend more on the efficiency with which the luminance signal is coded. Approximately speaking, for simple one dimensional coding of normal video-telephone type pictures, between 2 and 3 bits/pel might be considered adequate for luminance signal, whereas 1/2 to 1 bit/luminance pel is needed for both chrominance signals. Rubinstein [40] shows that DPCM using constant-length-code can achieve good quality with 2.81 b/pel for Q for use in different head and shoulder pictures; the reduction would be less for pictures with large amounts of detail.

It is not necessary to build a separate DPCM coder for the luminance and chrominance coder, [29] gives a simple method for using a single DPCM coder for the three components. DITEC [42], a digital

DPCM coder for the NTSC signal, operates at a bit-rate of 29.2 MB/s for the picture signal by DPCM coding with 5 b/pel for Y and 4 b/pel for I and Q, transmitted on alternate lines. The DPCM coder use uniformly spaced levels, switching between the fine spacing in low detail portion of the picture and coarse spacing at edges. One bit is used to indicate the scale being employed.

OCCITAN is a computer controlled coding for studying the DPCM coding of signal components derived from a SECAM signal, which requires 40 Mb/s to produce an acceptable picture quality for television transmission. As mentioned before, changes in the luminance signal partially mask the ability of the human observer to discriminate small perturbation in the chrominance signal in the immediate vicinity of the luminance changes. Having this in mind, design of luminance quantizer for DPCM has developed into quite an art [76]. The viewer's sensitivity to coding errors introduced at an edge are incorporated into the design procedure in different ways.

Plateau Coding

Plateau coding gives high-quality reproduction on single pictures, but when applied to a sequence of pictures, variations of the change points from frame to frame can produce disturbing effects [48].

Frame-to-Frame

In the frame-to-frame [24] coding technique, a circuit detects when there is a significant change between a reference frame memory and incoming picture signal. The frame-to-frame color should be adjusted to operate at transmission rates in the range 6.3 to 16 Mb/s.

At 6-3 Mb/s, large area movement, such as zooming and panning produce visible edge effects due primarily to the spatial subsampling of the signal (spatial subsampling is a control mode to prevent buffer overload). Buffer overload occurs for very active pictures, resulting in a certain amount of jitter. However, for video telephone or video-conferencing use, where the camera is fixed and there is little motion, the picture quality may be quite acceptable. At 16 Mb/s it is difficult to see degradation, and the picture quality is claimed to be approaching broadcast quality requirements.

Table V summarizes the different techniques for color video encoding.

TABLE V. Comparison of Different Techniques for
Color Video Encoding

Technique	Scene Variations	Performance	Implementation
PCM 3-D into 125 Volumes	Slowly Varing Scenes	40 Mb/s Some Color-Contouring is Visible	Relatively Simple
DPCM	Normal Conditions	29-9 Mb/s Good Results	Moderate
Plateau Coding	Still Pictures	Good Results 2.0 b/pel	Moderate
Frame-to-Frame Transform Encoding	Slowly Varing Scenes	6.3 Mb/s Acceptable Results	Complex with Memory and Buffer Overflow Problem
	Normal Scenes	16 Mb/s Good Results	

Current and Near Future Data Compression Systems

Model 100 is microprocessor-controlled voice digitizer, made by Time and Space Processing Inc. of Cupertino, California. The system uses linear predicting coding to convert a 4Khz analog voice into digital voice signal at only 2.4 Kb/s. The system permits two-way conversation in real time over standard phone lines using conventional telephones. It measures 18.6 by 16.7 by 18.8 inches, and sells for \$14,950 in small quantities. Electronics, August 4, 1977.

The German ITT subsidiary Standard Elektrik Lorenz AG., is testing a 34 Mb/s terrestrial digital TV link. Sending only the difference between adjacent picture elements on the scanning lines, using DPCM. The 34 Mb/s rate is for transmitting one complete color TV channel with two high-quality sound channels. After the signal has travelled to a satellite and back, it is demodulated and converted into analog form. On the internationally used 1-to-5 picture quality scale, the satellite transmitted picture scored between 4 and 5. Electronic, August 3, 1978.

U. S. Army is now flight-testing a video-compression module, at Fort Huachuca, Ariz. Built by RCA Corp., it is part of an early data link built by Harris Corp.. It divides a 256-line TV field into eight vertical stripes, each 32 pixels wide. The pixels are run through a 32 point discrete cosing transform and compared line by line using DPCM techniques. The system can transmit the TV signal at rates as low as 200 Kb/s.

A similar system with the same data rate is built by Data/Ware Development Inc., San Diego, Calif., for the Air Force. This system

is built with three 2901 bitslice microprocessor from advanced Micro Devices Inc.

Both of the above systems have good resolution, but the 256-by-256 Pixel image does not offer enough resolution for a pilot to track his target. Also, the picture tends to "tear" apart when the TV sensor is moving fast. Electronics, May 25, 1978.

The Army and Air Force have informally put together a joint program to develop video-compressor module, for various weapons-delivery systems. Two awards will go to contractors proposing systems that use one dimensional transform or time domain processing, or the combination of the two. The third will fund a two-dimensional transform approach using, for example, Hadamard transforms. Development and testing of the links should be completed by 1980-81. Electronics, May 25, 1978.

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CHAPTER V

CONCLUSIONS AND RECOMMENDATIONS

Different data compression systems have been presented and compared for various basic data sources. A true measure of overall compression efficiency has been defined as, "bandwidth compression". This is the ratio of the number of bits presented at the input to the number of bits delivered at the output of the data compressor, including all penalties for identification, timing, and synchronization.

For speech encoding, vocoders are the best candidate for 1980-1985 time-frame; these are specially formant vocoders because the system algorithm draws heavily on the results of recent research in linear predictive coding. There is need for future research in this area; because the complexity of vocoders can be overcome by use of large scale integrated circuits (LSI).

Specification for space vehicle telemetry systems must be determined largely from knowledge of previous data and from anticipation of future data behavior. However, the exploratory nature of space vehicles often produces data of unexpected behavior and any system used must cope with this and other possibilities at the expense of efficiency. Hence, it is desirable to have provisions to modify the data compressor performance in flight. It is generally considered a good design to use multiplexers, so one algorithm can be used for different data channels. This will enable the designer

to use a larger buffer, controlling the buffer fullness and having preprogramed the microprocessor for channel priority also. From the current state-of-the-art report one can extrapolate that large scale magnetic bubble memories will be commercially available, and use of LSI, and microprocessors give better on line control, which will result in better efficiency.

A discussion of buffer overflow is given in Chapter III (page 42, Implementation and Analysis). Note that all methods of controlling of buffer overflow will result in degradation of some or all the data, but in a controlled manner so that it is still useful .

For monochrome video coding interframe transform coding techniques (i.e. three dimensional transform coding with different block sizes ($16 \times 16 \times 16$), and ($4 \times 4 \times 4$), cosine and Fourier transform) seem to be the most promising approaches with good efficiency. However, extreme complexity with requirements for large storage, and long computations are associated problems; but, with the LSI's, which will be on the market in the near future, these problems can be overcome.

From the two color signal processing commonly used (RGB and composite) composite color video encoding with Y, I, Q components coded using DPCM, and run-length encoding are the more efficient and convenient approaches. Associated problems are large number of computations, large storage and buffer overflow; again these problems can be overcome as previously mentioned.

One approach that would find appeal among broadcasters, particularly in the near future, is to employ composite coding for distribution of signals within an area employing the same standard,

converting to component coding, when changing from one standard to another.

One result of data redundancy coding is the resulting need for error correcting coding, unless the data channel is highly reliable. Convolutional coding with Viterbi decoding algorithm [43] appears to be the most cost-effective forward error control technique. For more on this subject see references [17, 22, 23, 25, 26, 27, 28, 41, 43, 44, 47, 55, 57, 62].

APPENDIX I

BIBLIOGRAPHY

For the purpose of making a meaningful summary, the references are broken into parts: (1) data compression techniques for voice, (2) data compression for video signals. Included in the summary of each category will be a list of the most pertinent articles with a synopsis.

VOICE

The simplest form of digitizing speech is PCM, while PCM is widely used, it is not the most efficient technique for digitizing speech signals. DPCM results in better efficiency than PCM at the expense of slightly more complex implementation. Linear predictive algorithm and adaptive predictive coding has been used to digitize speech, resulting in better efficiency than PCM or DPCM. Vocoder using properties of linear predictive algorithms are the most promising method of bandwidth reduction for speech signals.

Following is a chronological list of some of the most pertinent articles related to data reduction for voice. A brief synopsis is included with each article in the list.

1. Ehrman, L., "Analysis of Some Redundance Removal Bandwidth Compression Techniques, "Proceedings of the IEEE, Vol. 55, No. 3, March 1967.

Theoretical performances of - the floating aperture prediction, the zero-order interpolation, and the fan-interpolator - are analyzed. Theoretical expressions are found for the mean and mean square times between output samplers of these devices; when the input signal is a Markov process.

2. Kortman, C. M., "Redundancy Reduction - A Practical Method of Data Compression," Proceedings of the IEEE, Vol. 53, No. 3, March 1967.

Techniques for accomplishing data compression, parameter extraction, adaptive sampling, predictors, and interpolators are discussed; examples of performances are given. Methods of achieving greater flexibility by combining redundancy reduction techniques with electronically programmable telemetry systems are discussed.

3. Ristenbatt, M. P., "Alternatives in Digital Communications," Proc. IEEE, Vol. 61, No. 6, June 1973.

A review of source encoding and methods for digitizing speech and video are summarized. A brief description of block and convolutional codes is given.

4. Lucky, R. W., "A Survey of the Communication Theory Literature: 1968-1973", IEEE Trans. on Information Theory, Vol. IT-19, No. 5, November 1973.

This paper surveys the literature on communication theory

from 1968 to 1973. Adaptive quantization topics are discussed initially, analog, and feedback communication are discussed.

5. Jacobs, I. M., "Practical Applications of Coding", IEEE Trans. on Information Theory, Vol. IT - 20, No. 3, May 1974.
Convolutional coding with soft-decision Viterbi decoding, and decoder implementations are discussed. Examples are given.
6. Noll, P., "A Comparative Study of Various Quantization Schemes for Speech Encoding", The Bell Syst. Tech. J., November 1975.

Performance limits, given in signal-to-noise ratio are discussed for adaptive quantization, and linear prediction schemes. Examples of performance are given.

7. Amcheta, T. C., Jr., "Syndrome-Source-Coding and its Universal Generalization", IEEE Trans. on Information Theory, Vol. IT - 22, No. 4, July 1976.

A method of using error correcting codes to obtain data compression is described. A "Universal" generalization of this method is formulated which provides robustly effective distortionless coding of source ensembles.

8. Communication, "Voice System Saves Bandwidth", Time and Space Processing Inc. 10430 Tantau Ave., Cupertino, Calif. 95014 from: Electronics, August 1977.

Commercially available equipment for linear predictive coding for digitized voice channels for telephone grade channels of 2.4 Kb/s is reported.

9. Gibson, J. D., "Sequentially Adaptive Backward Prediction in ADPCM Speech Coders", IEEE Trans. on Commun. Vol. COM-26, No. 1, January 1978.

Using a Jaynt-type adaptive quantizer, it is shown that for bit rates less than 16 Kb/s with second order predictors and for bit rates less than 18.4 Kb/s with fourth order predictors, backward adaptive predictors have a definite advantage over fixed-tap predictors, since the latter may cause system divergence.

10. Un, C. K., "A Low Rate Digital Formant Vocoder", IEEE Trans. on Commun., Vol. COM-26, No. 3, March 1978.

A complete algorithm of 1200 bits/s digital formant vocoder is described. This vocoder algorithm draws heavily on the results of recent research in linear predictive coding.

Other references which are related to data compression schemes for speech are listed below. Numbers indicate those entries in the bibliography in Appendix I. 3, 4, 6, 7, 8, 10, 12, 13, 15, 16, 20, 21, 22, 30, 31, 32, 38, 44, 46, 47, 49, 51, 52, 53, 54, 57, 59, 63, 64, 77, 80, 83.

VIDEO SIGNALS

Data Compression for video signals are based on the properties of the vision of human observer; psychological coding are also used. Video signals can be divided into monochrome, and color video signals. For monochrome frame to frame picture coding, then using run-length coding will give efficient coding. Color video signals are coded in either component RGB or composite signal coding using PCM, DPCM, or interframe coding techniques. For color video composite signal using DPCM Coding, then using run-length coding results in efficient coding.

Following is a chronological list of some of the most pertinent articles related to data reduction for video signals. A brief synopsis is included with each article in the list.

1. Schreiber, W. F., "Picture Coding", Proceedings of the IEEE, Vol. 55, No. 3, March 1967.

Consideration of picture coding with regard to human vision; and relationship between different quantization and PCM is discussed in detail.

2. Candy, J. C., Franke, M. S., Haskell, B. C., and Mounts, F. W., "Transmitting Television as Clusters of Frame-to-Frame Differences", Bell Syst. Tech. J., Vol. 50, July-August 1971.

Minimizing the visibility of granular coding noise, nonlinear properties of human visual system, and statistics of the source are three topics which have been dealt with in some detail.

3. Taylor, W. E., Jr., C. L. Walker, and F.M. Ingels, "An Investigation of Compact Codes for the Geometric, Poisson and Binomial Probability Distributions," Engineering and Industrial Research Station, Electrical Engineering, Mississippi State University, EIRS-EE-73-1, July 1972.

A detailed study of the average code word lengths versus the encoded word lengths for run-length encoding of sources with Geometric, Poisson, Binomial and Modified Binomial source distributions is given.

4. Roese, J. A., Pratt, W., and Robinson, "Interframe Cosine Transform Image Coding," IEEE Trans. on Commun., Vol. COM-25, No. 11, November 1977.

Three-dimensional cosine transform coding, and two-dimensional cosine transform coding with an image frame combined with DPCM coding between frames are analyzed. Simulation results are presented.

5. Haskell, B. G., Gordon, P. L., Schmidt, R. L., and Scattaglia, J. V., "Interframe Coding of 525-line, Monochrome Television at 1.5 Mb/s", IEEE Trans. on Commun., Vol. COM-25, No. 11, November 1977.

Interframe coder for TV-conferencing or video telephone, where the television camera is largely stationary is presented. A simulation of the results is included.

6. Limb, J. O., Rubinstein, C. B., Thompson, J. E., "Digital Coding of Color Video Signals- A Preview", IEEE Trans. on Commun., Vol. COM-25, No. 11, November 1977. Basic definitions of color video signals and different coding scheme for color video is given. This paper is a complete review for color image coding.

7. Musman, H. E., and Preuss, D., "Comparison of Redundancy Reduction Codes for Facsimile Transmission of Documents", IEEE Trans. on Commun., Vol. COM-25, No. 11, November 1977.

A comparison and evaluation of different redundancy reduction coding techniques (one-dimensional, and two-dimensional coding) for digital transmission of facsimile documents on telephone lines presented.

8. Zschunke, "DPCM Picture Coding With Adaptive Prediction", IEEE Trans. on Commun., Vol. COM-25, No.11, November 1977.

An effective way for determining the contour direction, taking into account signal-source noise, quantization, distortion, and hardware complexity is given. A simulation of test signal of broadband TV picture is presented.

- .9. Mark, J. W., and Daslewiz, P. P., "Reference Encoding of Correlated Digital Source", IEEE Trans. on Commun. Vol., COM-25, No.11, November 1977.

A reference code, with an associated bound on the code rate,

for the encoding of correlated digital sources is introduced.

The reference code is adaptive and simple to implement.

10. Sawada, K. and Kotera, H., "A 32-Mbits/s Component Separation DPCM Coding System For NTSC Color TV", IEEE Trans. on Commun., Vol. COM-26, No.4, April 1978.

A component separation DPCM for NTSC color TV signal is first separated into a luminance component Y and two chrominance component I and Q, the system described is capable of transmitting a 4 MHz NTSC color TV Signal with broadcast quality at 32.064 Mb/s rate.

Other references which are related to data compression schemes for video signal coding are listed below. Numbers indicate those entries in the bibliography in Appendix I: 1, 2, 5, 9, 10, 11, 18, 19, 24, 29, 33, 34, 35, 36, 37, 39, 40, 45, 48, 49, 50, 51, 52, 56, 60, 61, 63, 65, 66, 67, 68, 69, 70, 71, 72, 73, 74, 75, 76, 78, 79, 81, 82, 83.

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